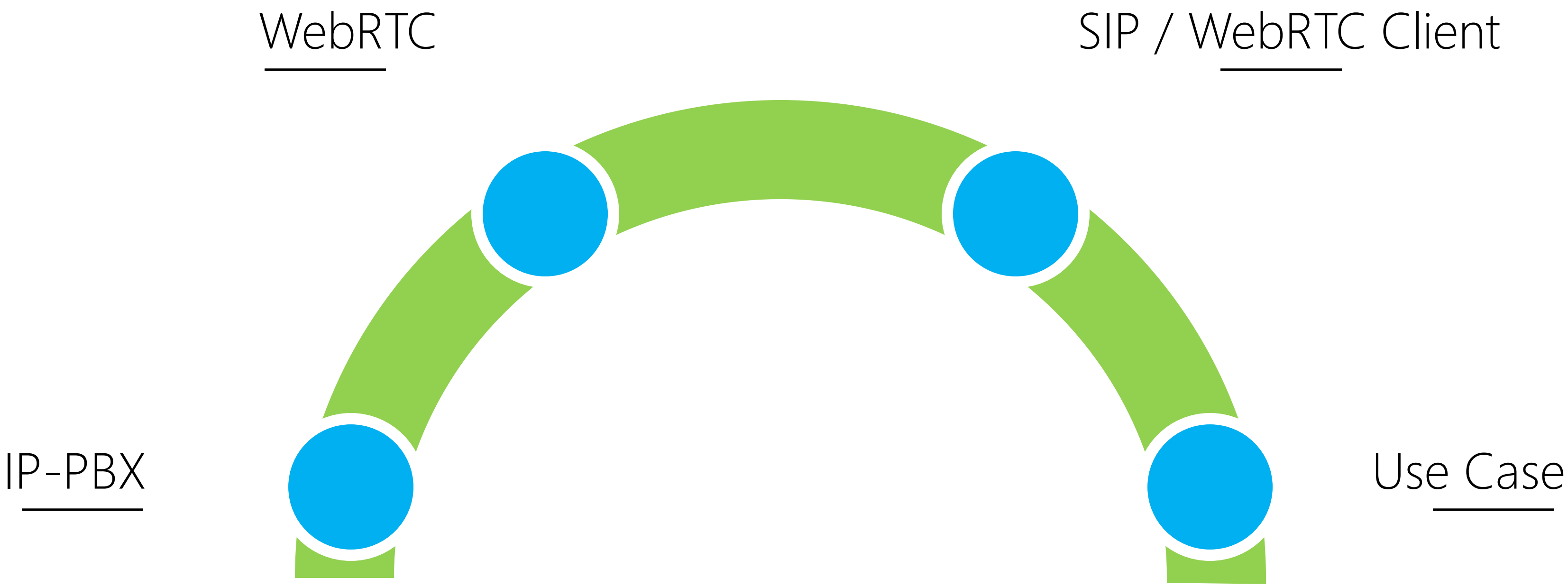
Vietnam Free & Open Source Software Association

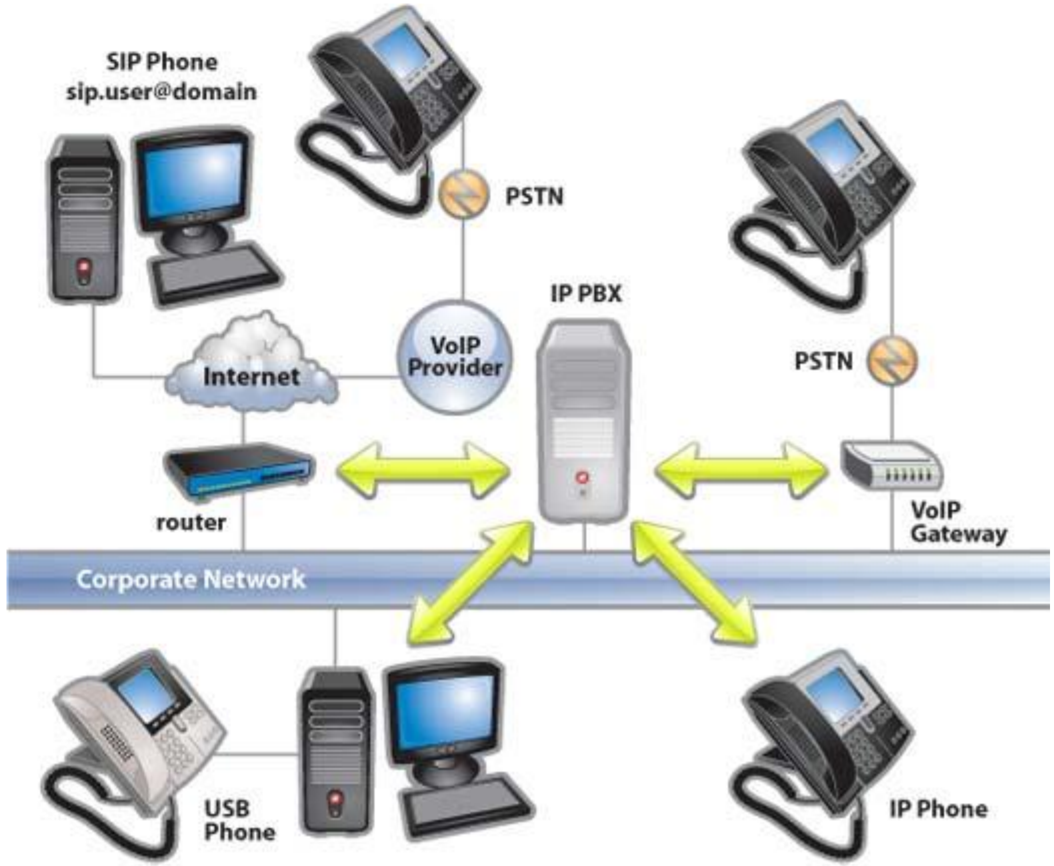
Open Telecom Application Platform for Startups

Au Duong Dat, Chairman - VHT

Content



1 IP-PBX



PBX (Private Branch Exchange) is a system that connects telephone extensions to the **PSTN** (**Public Switched Telephone Network)** and provides internal communication for a business. **An IP-PBX** is a PBX with Internet Protocol connectivity and may provide additional audio, video, or instant messaging communication utilizing the **TCP/IP** protocol stack.

Asterisk is an open source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. **Asterisk** powers IP PBX systems, VoIP gateways, conference servers and other custom solutions. It is used by small businesses, large businesses, call centers, carriers and government agencies, worldwide. **Asterisk** is free and open source. **Asterisk** is sponsored by **Digium**

FreeSWITCH is a scalable open source cross-platform telephony platform designed to route and interconnect popular communication protocols using audio, video, text or any other form of media. It was created in 2006 to fill the void left by proprietary commercial solutions. FreeSWITCH also provides a stable telephony platform on which many applications can be developed using a wide range of free tools. FreeSWITCH was originally designed and implemented by Anthony Minessale II with the help of Brian West and Michael Jerris. All 3 are former developers of the popular Asterisk open source PBX.

2

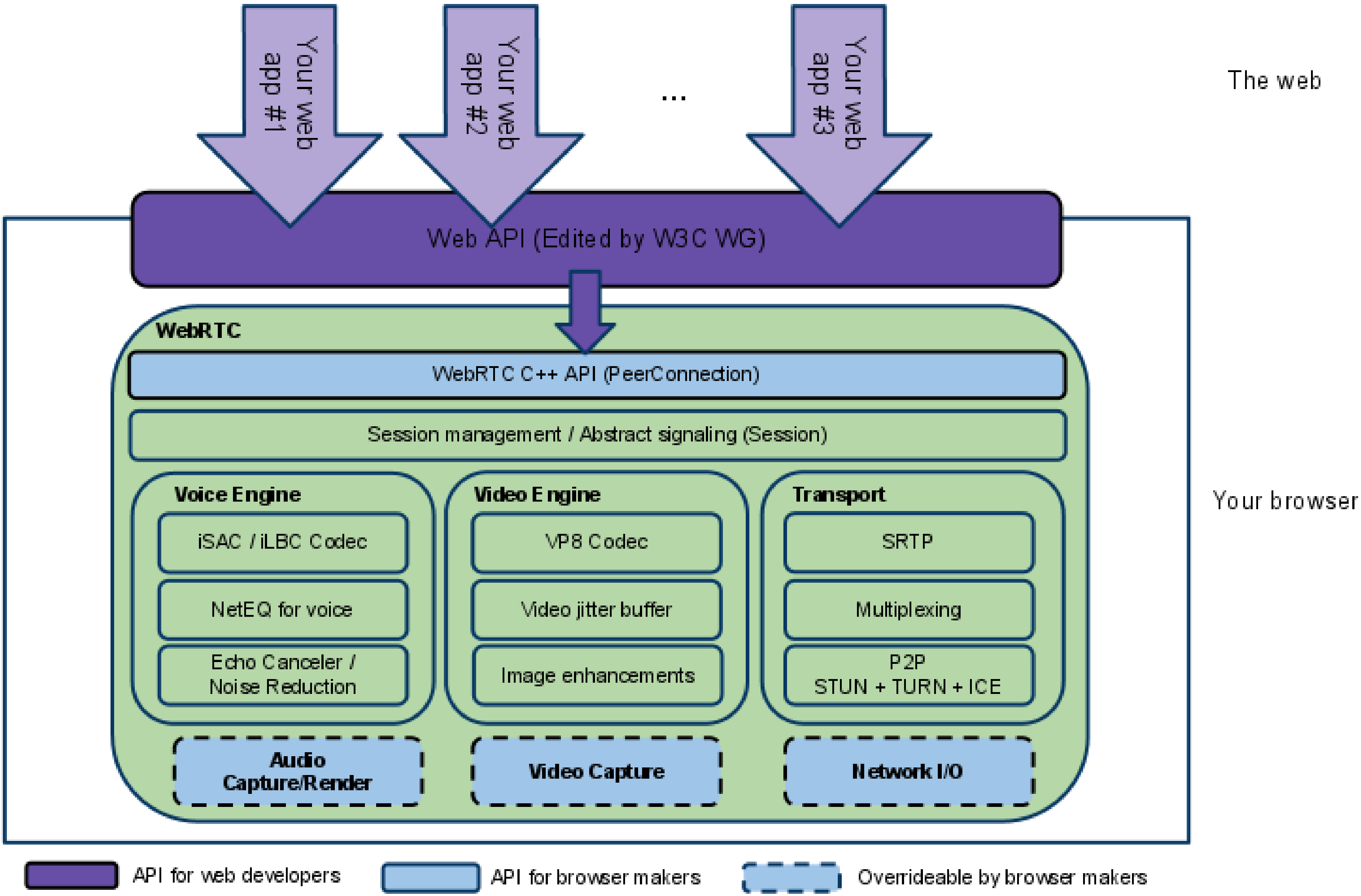
WebRTC

WebRTC is a free, open project that provides browsers and mobile applications with Real-Time Communications (RTC) capabilities via simple APIs. The WebRTC components have been optimized to best serve this purpose.

WebRTC enable rich, high quality, RTC applications to be developed for the browser, mobile platforms, and IoT devices, and allow them all to communicate via a common set of protocols.

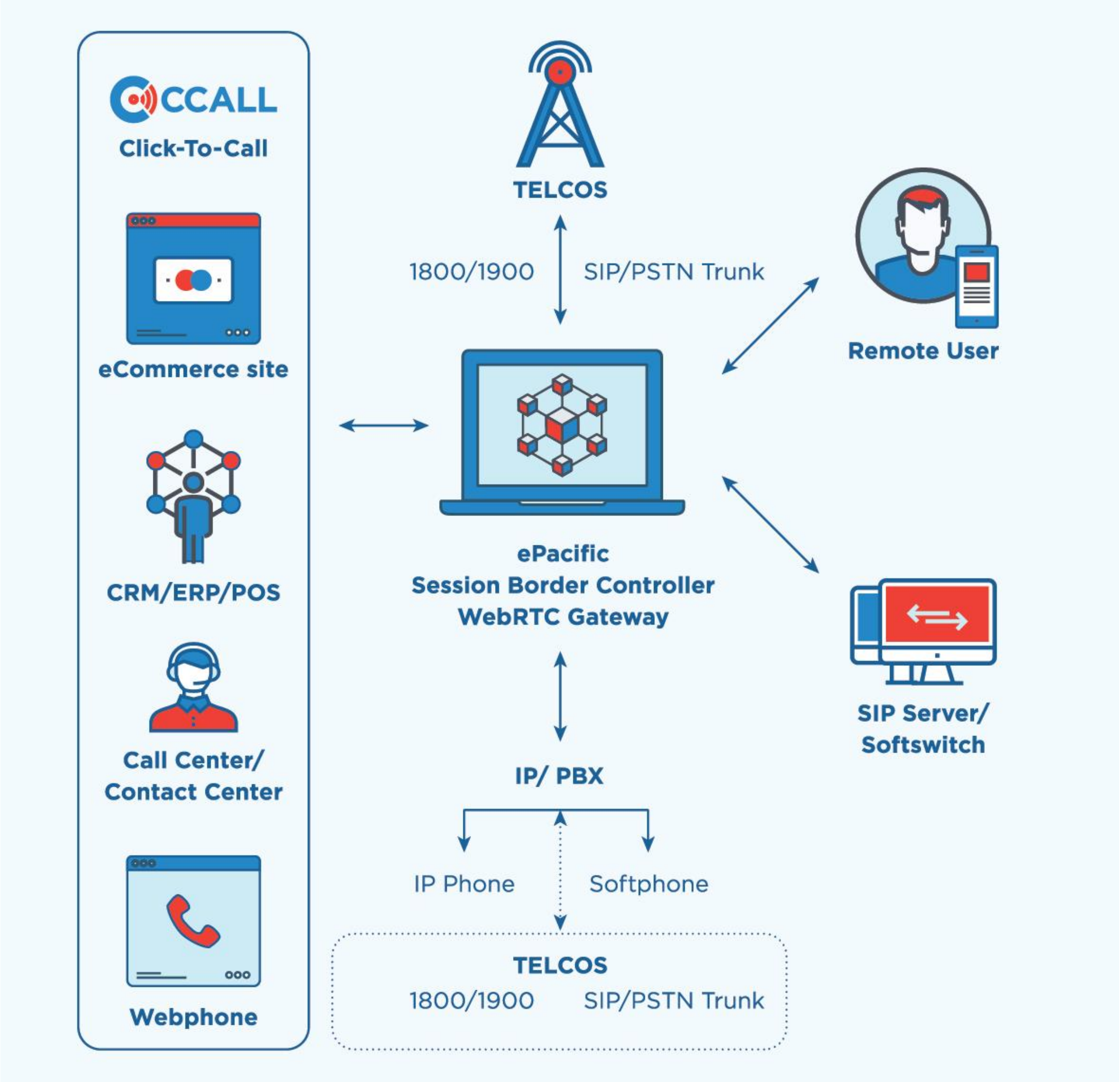
The **WebRTC** initiative is a project supported by Google, Mozilla and Opera, amongst others.

WebRTC Architecture

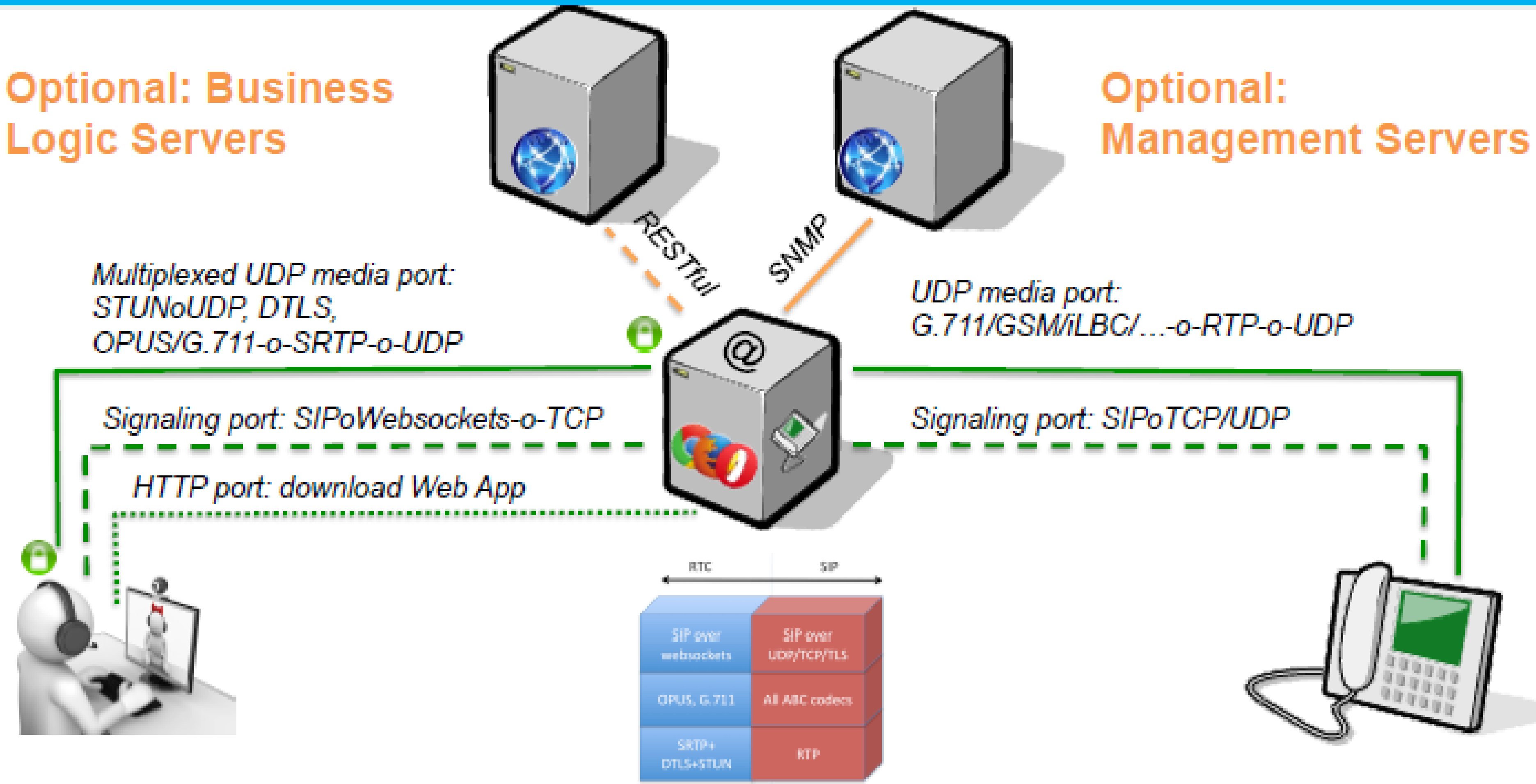


3 SIP / WebRTC Client

WebRTC Gateway



Protocol Reference Diagram



WebRTC Browers

ePacific WebRTC Gateway

SIP Equipment

JsSIP

JsSIP is a simple to use **JavaScript** library which leverages latest developments in **SIP** and **WebRTC** to provide a fully featured SIP endpoint in any website. With **JsSIP** any website can get Real Time Communications features using audio, video and more with just a few lines of code.

SIP over WebSocket transport.

Audio/video calls, instant messaging and presence.

Lightweight!.

100% pure JavaScript built from the ground up.

Easy to use and powerful user API

Getting Started

JsSIP User Agent is the core element in JsSIP. It represents the SIP client associated to a SIP account. JsSIP User Agent is define
`JsSIP.UA` class.

Multiple JsSIP User Agents can be created (this is useful for having different SIP accounts running in the same web application).

Creating a JsSIP User Agent

User Agent Configuration

JsSIP User Agent requires a configuration object for its initialization. There are some mandatory configuration parameters and m
optional ones. Check the full configuration parameters list.

```
var configuration = {
  'ws_servers': 'ws://sip-ws.example.com',
  'uri': 'sip:alice@example.com',
  'password': 'superpassword'
};
```

User Agent instance

```
var coolPhone = new JsSIP.UA(configuration);
```

User Agent events definition

Full list of User Agent events.

WebSocket connection events

```
coolPhone.on('connected', function(e){ /* Your code here */ });

coolPhone.on('disconnected', function(e){ /* Your code here */ });
```

New incoming or outgoing call event

```
coolPhone.on('newRTCSession', function(e){ /* Your code here */ });
```

New incoming or outgoing IM message event

```
coolPhone.on('newMessage', function(e){ /* Your code here */ });
```

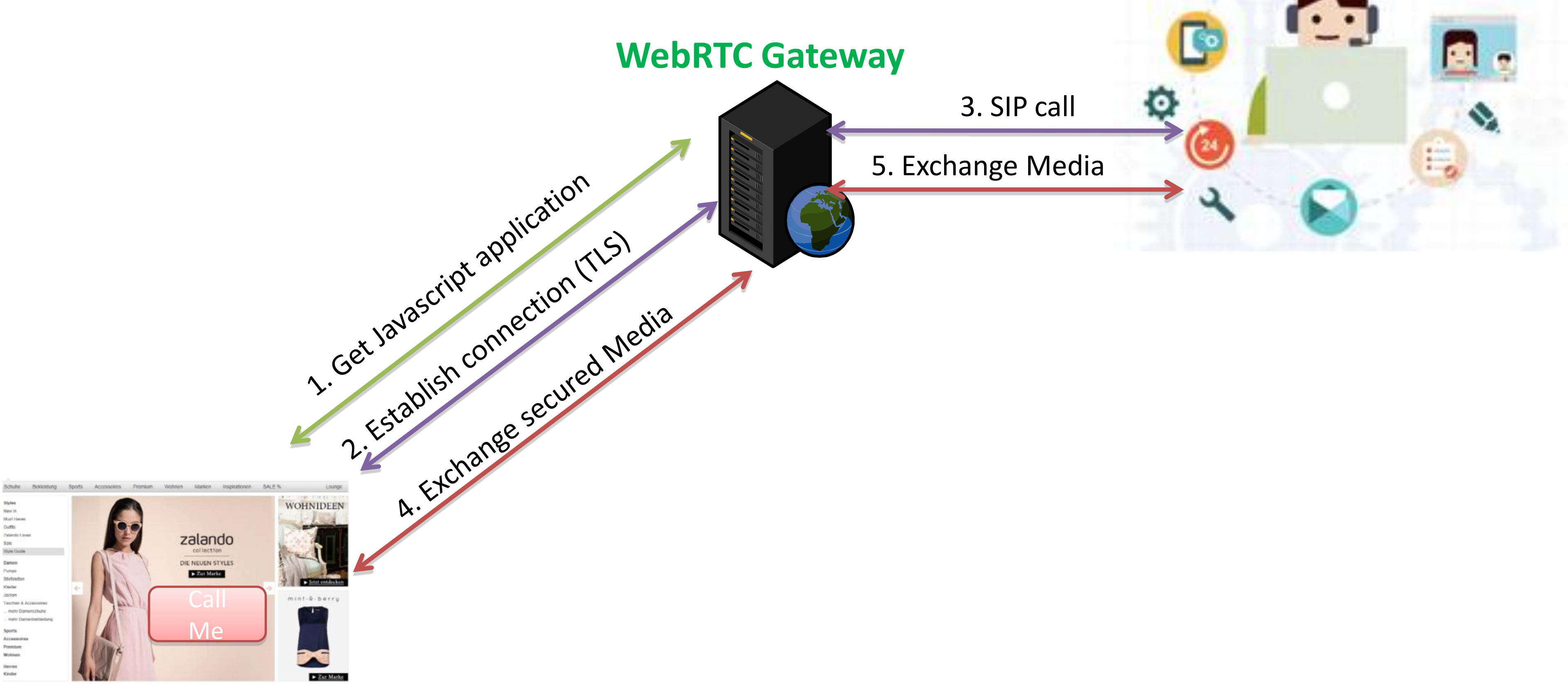
SIP registration events

```
coolPhone.on('registered', function(e){ /* Your code here */ });
coolPhone.on('unregistered', function(e){ /* Your code here */ });
```

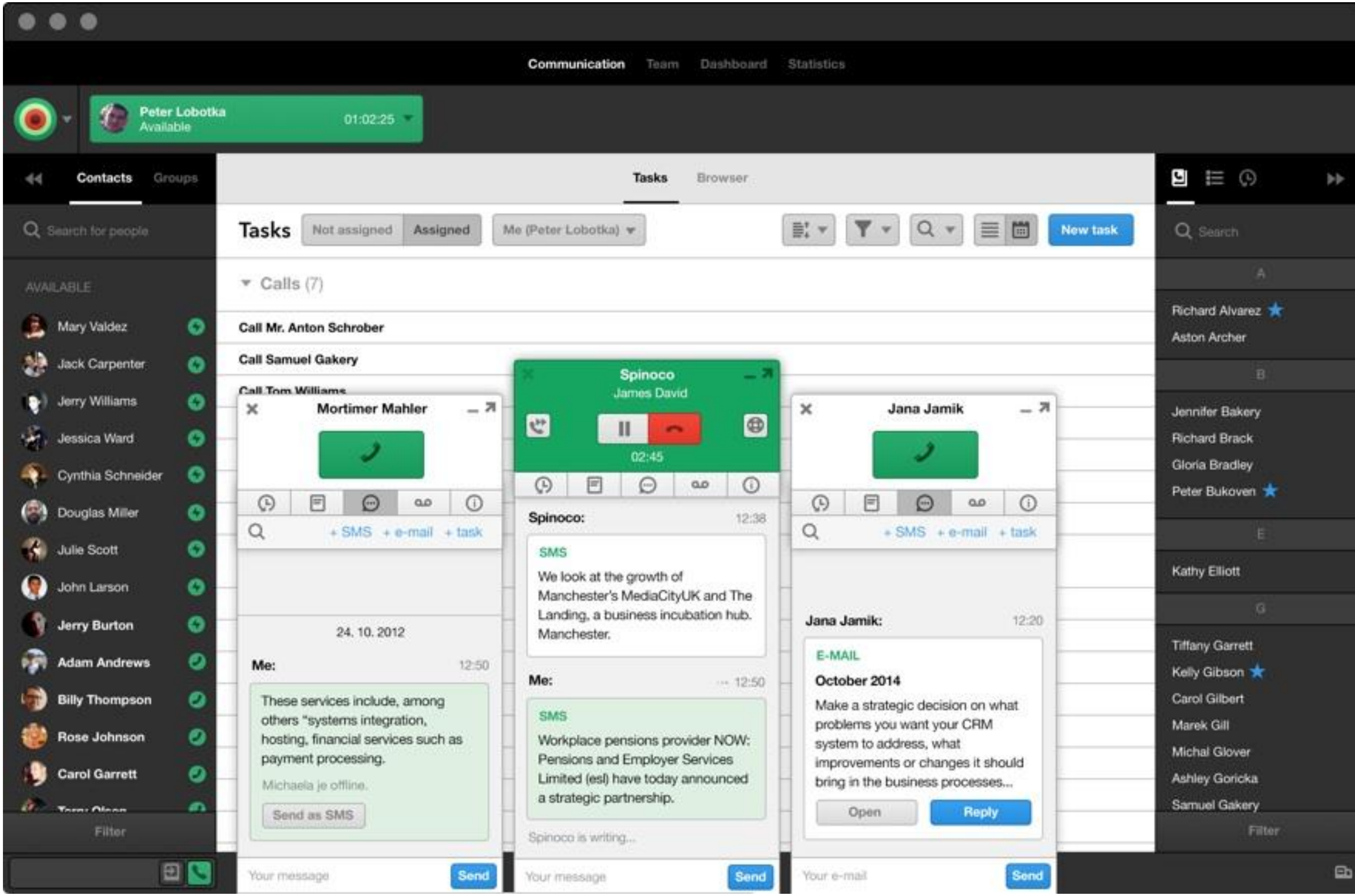
4 Use Cases

Online Customer Interaction

1. Users download the application as a Javascript
2. Caller establishes a call with the WebRTC gateway
3. WebRTC gateway established a VoIP call to Call center
4. Caller and Call center exchange data

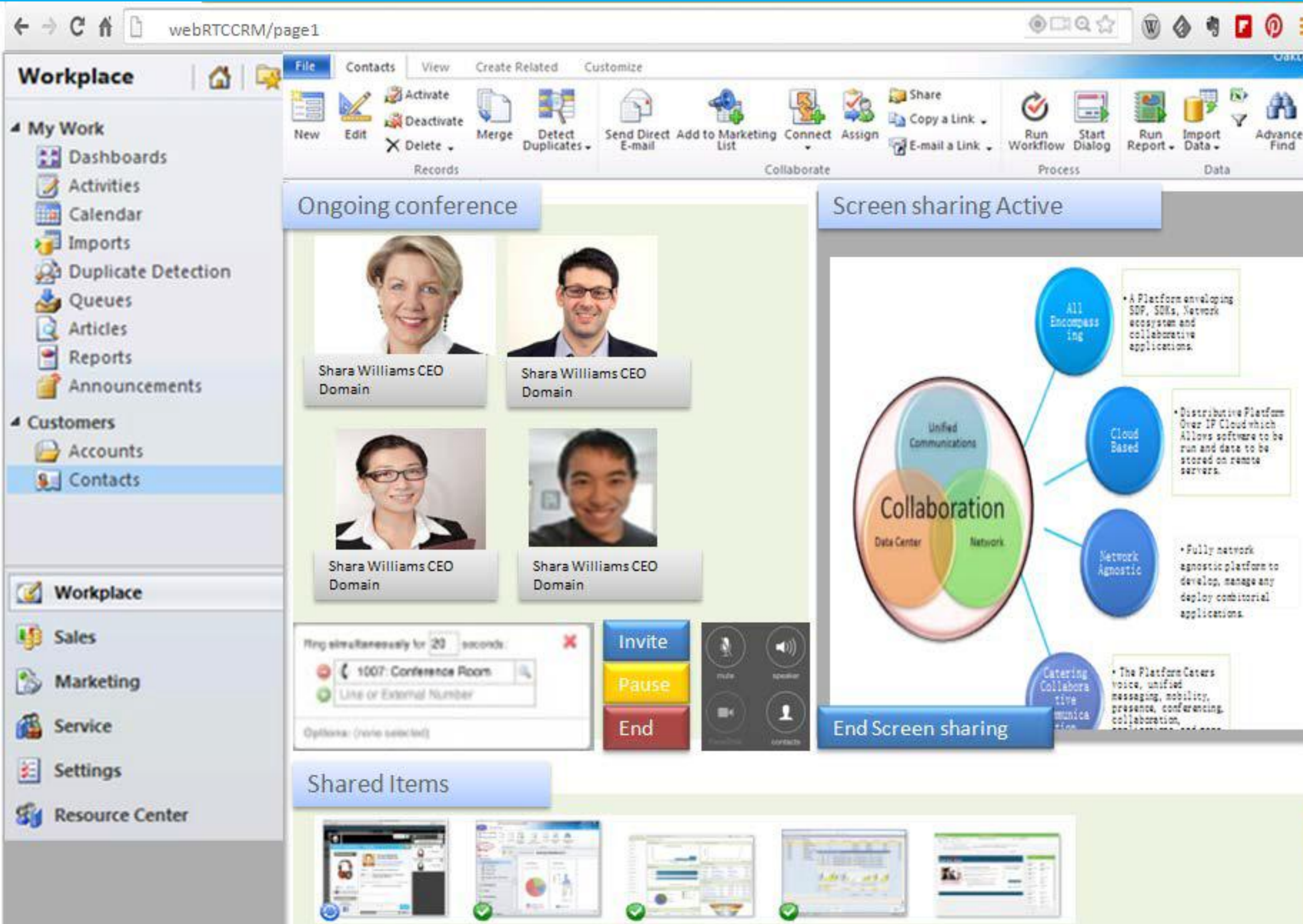


Contact Center Software

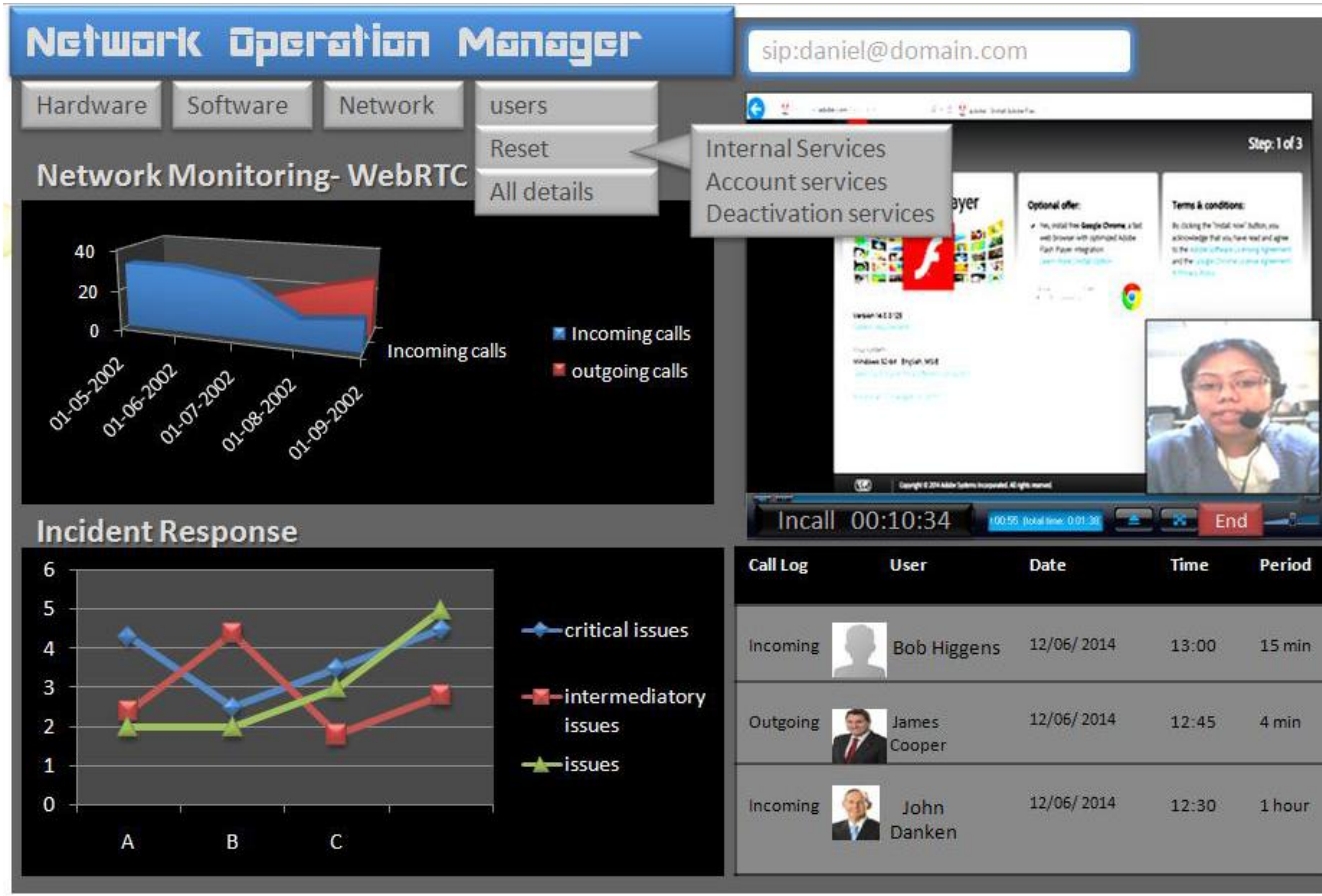


Team Communicator

Customer Relationship Software



Network Operation Center



Human Resource Management

JOBS online

Java sip webrtc | Find

ALTANAI BISHT

Objective

Maverick software engineer aspiring to utilize her technical and interpersonal skills while being innovative .

Personal Profile

Date of Birth	22/09/1989
Sex	Female
Father's name	D.S Bisht
Mother's name	Maya Bisht
Languages known	English , Hindi , Japanese

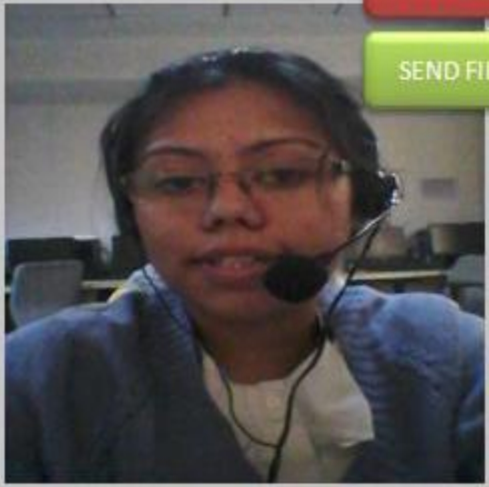
Skills

Language	: C, C++, C# , VisualBasic ,JAVA (NIIT certified , SCJP certification), J2EE(NIIT certified), JavaME , Microsoft.net , XML , HTML 5
Scripting	: Javascript , PHP, SQL ,unix shell , ASP , JSP, AJAX , CSS 3
Packages	: MS-Office, Oracle 10 ,Visual programming , Netbeans , Eclipse , IBM Rad ,Salesforce CRM
Platforms	: Linux ,Windows 95/98/2000/XP/Vista/7, Symbian , bada , android
Networking	:WAN technologies(T1/DS1,T3/DS3,ISDN), Intranet/Internet technologies andprotocols (VOIP , DNS , SMTP , POP3 , HTTP , TCP/IP , SSH , ATM , TCP/IP , VPN)

HOLD

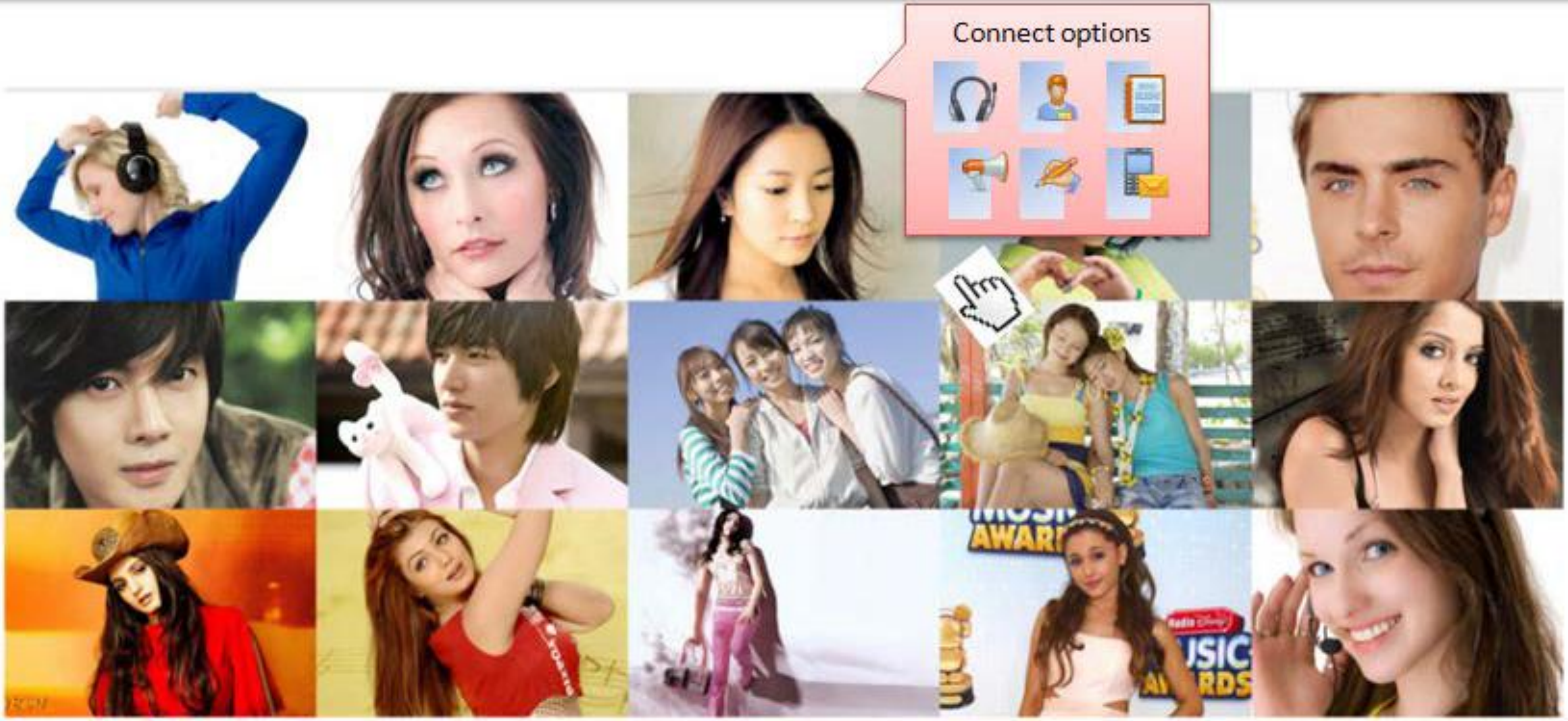
END CALL

SEND FILE



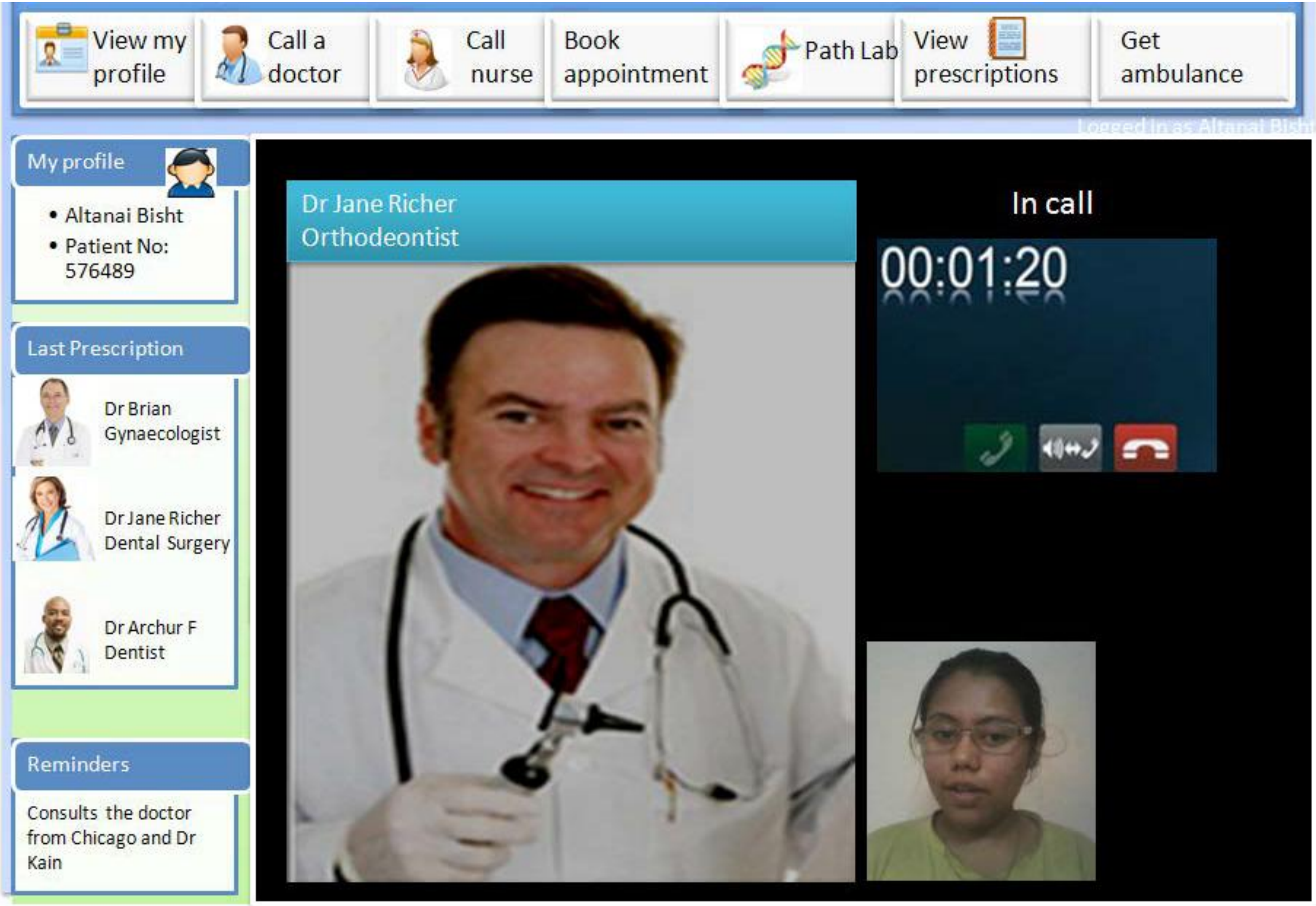
In call 00:10:45

Dating Site

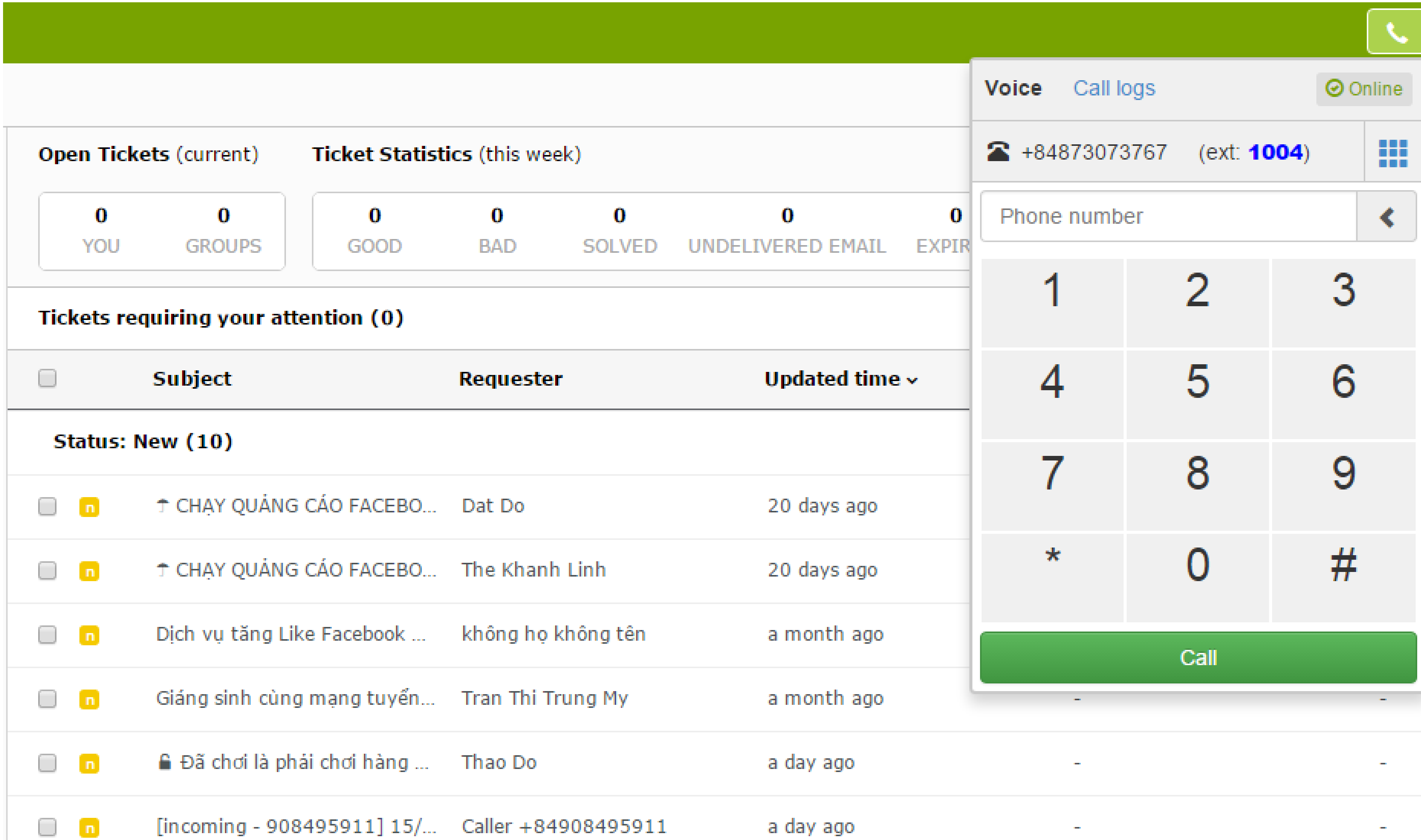


Call Log		User	Date	Time	Period
Incoming		User394889	12/06/ 2014	11:00	5 minute
Outgoing		User297923	12/06/ 2014	12:45	4 minute
Incoming		User093898	12/06/ 2014	19:30	1 minute

Online Medical Consultation



Izi Help



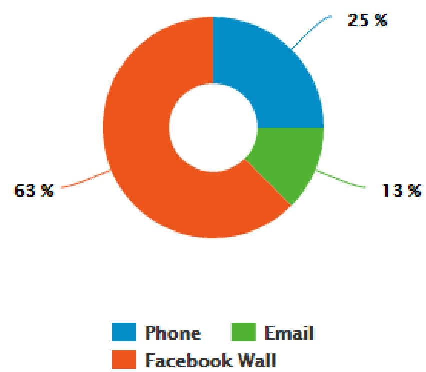
Voice
Call logs
Online
+84873073767 (ext: 1004)
Phone number
1 2 3
4 5 6
7 8 9
* 0 #
Call
Open Tickets (current)
Ticket Statistics (this week)
0 YOU
0 GROUPS
0 GOOD
0 BAD
0 SOLVED
0 UNDELIVERED EMAIL
0 EXPIR
Tickets requiring your attention (0)
Subject
Requester
Updated time
Status: New (10)
CHẠY QUẢNG CÁO FACEBO... Dat Do 20 days ago
CHẠY QUẢNG CÁO FACEBO... The Khanh Linh 20 days ago
Dịch vụ tăng Like Facebook ... không họ không tên a month ago
Giáng sinh cùng mạng tuyển... Tran Thi Trung My a month ago
Đã chơi là phải chơi hàng ... Thao Do a day ago
[incoming - 908495911] 15/... Caller +84908495911 a day ago

Izi Help

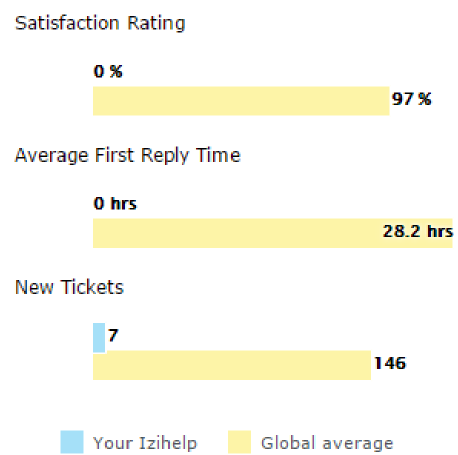
Ticket Stats

7	0	6	1	0%	0 hrs
New Tickets (total)	Solved Tickets (total)	Backlog (average)	Agent Touches (total)	Satisfaction Rating (average)	First Reply Time (average)

Tickets by Channels



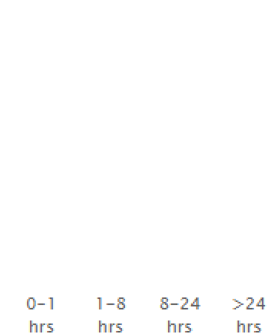
Benchmark



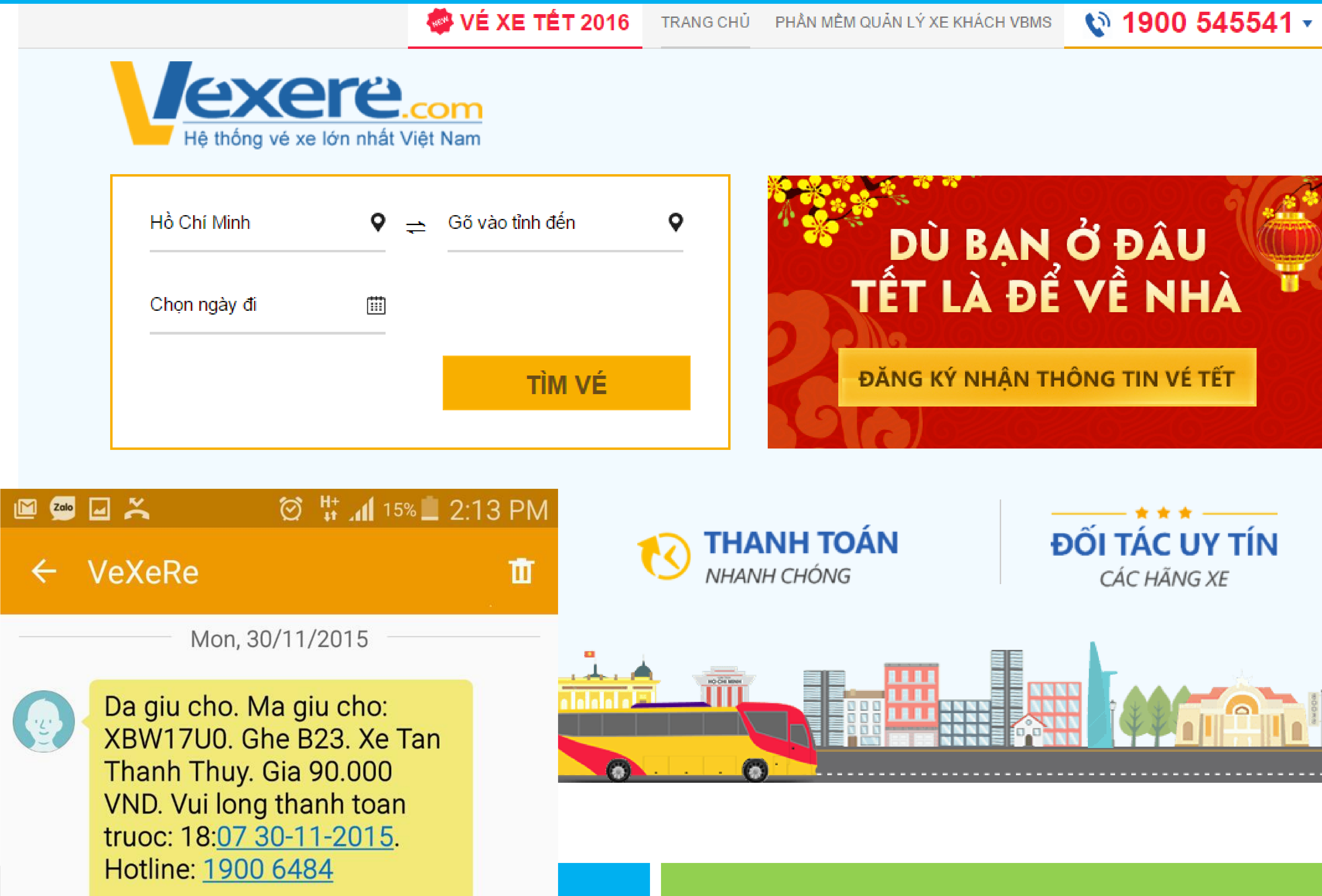
First Reply Time

0 hrs

Avg. First Reply Time



VeXeRe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ĐĂNG KÝ NHẬN THÔNG TIN VÉ TẾT
THANH TOÁN
NHANH CHÓNG
ĐỐI TÁC UY TÍN
CÁC HÃNG XE
15% 2:13 PM
VeXeRe
Mon, 30/11/2015
Da giu cho. Ma giu cho: XBW17U0. Ghe B23. Xe Tan Thanh Thuy. Gia 90.000 VND. Vui long thanh toan truoc: 18:07 30-11-2015. Hotline: 1900 6484


T H A N K Y O U