VoipSwitch User Portal for Rich Communication Suite
RCS features, HTML 5, WebRTC powered
FOR DESKTOP AND MOBILES
The VoipSwitch User Portal (VUP) is a self-care customer portal for VoIP service providers. Its functionalities combine the RCS features with account management and billing facilities.

The design is contacts oriented with quick access to Contacts stored on the server in the Network Address Book and shared across all RCS devices (native clients for smartphones, tablets and desktops). The RCS contacts are presented with their presence status and social profile details.

The history is shared across all clients and is archived on the server. Wherever a user logs in from he will get the latest communication details.

Quick access to handy widgets responsible for various functionalities is provided through the dashboard. A user can set Find me, browse voicemail, send fax, initiate Chat etc.

The calling features are realized using the WebRTC media engine which guarantees an unrivaled voice and video quality experience directly from a browser*. The approach taken in the technology behind WebRTC corresponds to the efforts VoipSwitch undertook in recent developments in its network elements (softswitch and application servers) and SIP clients such as full support for ICE framework, transcoding for breakout calls for the OPUS codec, support for secure RTP, adding new codecs to Voicemail and IVR modules, handling VP8 video and others.

Thus the VoipSwitch platform has become a perfect enabler for WebRTC based communication. Users can make calls directly from a browser to VoIP native applications built for various operating systems and yet benefit from the use of peer to peer communication, perfectly clear OPUS coded audio calls and fidelity of VP8 video.
Architecture

The portal front end is built using the HTML 5 standard with JavaScript as a client responsible for communication with the back end infrastructure.

All commands resulting from a user’s actions like sending a message, changing the presence status or initiating a call are sent to the WebToSIP server through Web Socket.

WebToSIP is a server component responsible for translating commands to SIP requests and forwarding them to the SIP softswitch (Voipswitch). Also, it handles all responses and requests coming from the SIP side and converts them to commands that can be understood by the portal JS client and passes them through the web channel.

It is important from the security point of view that the login credentials used for logging in to the portal are not the SIP ones. The SIP details are never revealed to the end user.

During logging in the WebToSIP server communicates securely with the authentication API from where it receives the SIP parameters used in further communication with the SIP server. The credentials are kept only until the session with the client is active.

The SIP server is a class 5 softswitch; it can be VoipSwitch or any 3rd party softswitch solution.

The requirements are that the softswitch has to support ICE and also pass certain parameters in SDP which are connected to sRTP. Also, it has to allow OPUS and VP8 codecs (for audio and video respectively).
Another required network element is TURN server. VoipSwitch provides its own TURN solution which support the latest standards including support for both UDP and TCP. TURN is necessary for NAT traversal and in the ICE framework which allows peer to peer calling with fallback to relay through server.

ICE is supported on WebRTC and also across all our RCS native apps. This framework ensures that users connect to a service from any network, especially if enhanced by TCP relay as the last resort relay when UDP is blocked.

There is one more aspect which has to be taken into account, namely sending calls to a breakout gateway. WebRTC implementation in browsers requires that audio is encrypted (sRTP). As the offnet traffic is sent to carriers which expect RTP, the traffic has to be decoded beforehand.

VoipSwitch’s transcoding application server provides a facility for encrypting/decrypting calls as well as encoding/decoding from for example OPUS to g729 or any other codec commonly used in interconnections.

More on ICE can be found in the following article:
http://www.jdrosen.net/uploads/1/5/0/0/15008848/ice-basic-tutorial.pptx
RCS features

High quality audio and video calling

The portal allows users to make both ONNET (calls to other RCS clients) and OFFNET (to non RCS numbers) calls.

ONNET calls can be initiated directly from Contacts after selecting an RCS user or from a keypad component which is one of the dashboard widgets. When dialing a number from the keypad it checks with the RCS API if the number is an RCS user; if so, it will try to connect ONNET.

The audio calls use the OPUS codec which proved to be a brilliant choice for VoIP and is now being widely adopted by the main service providers. VoipSwitch native clients, voicemail server and the transcoding module are also fully compliant with OPUS. In short, the codec combines what is best from SILK with audio preprocessors and dynamic adaptive technologies resulting in perfect voice quality and very efficient resiliency against bandwidth congestions.

Video calls use the VP8 codec which, thanks to Google initiatives, is becoming a standard in video services, gaining over its rival h264.
RCS features

Contacts - Network Address Book

The portal is integrated with the RCS Network Address Book which shares contacts from multiple user devices. A user's contacts from his iphone or android smartphone (only those which have been entered from the app or those which are RCS) will appear automatically in NAB; also contacts entered in the portal will be shared across devices.

The contacts are shown in the form of tiles. RCS contacts are differentiated by showing them in different colors representing presence statuses and also with avatars. Clicking on a particular RCS contact shows the profile details.

The “Find friends” option allows searching for other RCS users by name, email address or a service ID (username).
RCS features

Instant messaging (Chats)

This functionality utilizes the SIP SIMPLE framework for instant messaging. The translation to SIP takes place on the WebToSIP server. Messages history is shown in much the same way as is common now on mobile devices, i.e. divided in threads per addressee. Clicking on a particular thread opens a conversation window showing messages from and to the user (chat).

Chat can be started from the Contacts menu by clicking on the chat icon of a chosen RCS user. It can also be opened from the Chats widget on the Dashboard.

The portal also supports sending OFFNET messages which are routed through the SIP softswitch to an SMS carrier and delivered to mobile devices.

Social Profile Information

The My profile menu aggregates user’s personal details which are then publicly available and visible to other RCS users. Some of them are used in the Find friends option and allow finding a user by different criteria.

The Profile picture (avatar) is what other users see in the contact detail view.

A user can also set his status text (so called “free text”) which is then published to users which have him added in contacts.

Sharing a presence status and profile picture is based on standard SIP Presence with support for Resource lists (RCS users list). Translation to SIP is done on the WebToSIP server which subscribes to the list on behalf of the user logged in to the portal.
Other functionalities

Signup

Built in sign up component. This incorporates email address verification. After submitting a form on the web the user receives an email with a verification link. Clicking on the link creates an account and the user logs in to the Portal.

Notifications

Shows new events such as an incoming call, new chat message, new voicemail. It is shown on the toolbar.

Phone numbers (DIDs)

This functionality enables users to order geographical phone numbers (DIDs). The backend system can be connected to major DIDs providers’ APIs (e.g. Voxbone) which allows dynamic DID assignment. It supports a local DID database as well. A user can choose the country and city/region of the DIDs. All incoming calls and SMSes (subject to provider supporting SMS) are delivered to an RCS user. Also, a user can set his DID as the callerID for OFFNET calls. Payments for DIDs can be periodical (monthly, quarterly) or one time.

Shared history

This menu shows all recent activity of the user, from all his/her devices. It aggregates chat messages, calls and voicemails.

In addition there are separate tabs dedicated for a particular kind of activity (calls, messages) and showing them in greater detail.

The history can be also filtered and exported to a csv file.
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Other functionalities

USER’S PROFILE

A menu in which a user can manage his account, both the public social profile and also his billing part and the services settings.

PERSONAL DATA

Social profile, information grouped here is publicly available to other RCS users and is used in the Find friends user directory.

INVOICES AND PAYMENT HISTORY

ACCOUNT

SERVICES

For more information please contact: sales@voipswitch.com

www.voipswitch.com
Other functionalities

DASHBOARD

Dashboard is a launchpad for various widgets. Below are the ones that come with the default package:

FIND ME

Find me – a user can set rules for behavior when a call is rejected (either by the user or cannot be delivered for some other reason), depending on who is calling and to which phone number/DID. The actions include call forwarding to another phone number or to voicemail.

In addition, a voicemail greeting and the greeting being played while connecting can be personalized depending on the caller.

SEND FAX

Send fax – sending faxes and quick look at fax history.

CHAT

Chat – chats with other users.

VOICEMAILS

Voicemails – list of received voicemails and quick configuration of the voicemail account.

DO NOT DISTURB

Do not disturb – a user can list the caller IDs for which he will not be available for calls; he can also set the option ANY, which means he is not available to anybody.

DIALPAD

- Calls – user can dial a number and make a call; if the user is an RCS user then the appropriate information will be shown, otherwise the rate per minute will appear.

- SMS – same as above but for short messages; if a user is an RCS user it will be sent as chat message, otherwise it will be converted to an SMS on the server side and delivered further to a GSM user.

BALANCE TRANSFER

Balance transfer – a user can send credit from his balance to another RCS user.

INTERNATIONAL MOBILE TOPUP

International mobile topup – allows topping up GSM accounts; works with selected GSM carriers. The aim is that an RCS user can send some of the credit from his balance to a non RCS user.