VNC
Virtual Network Consult

VNCtalk
for Zimbra

Requirements Document

Softphone and SIP connection for VNCtalk

Rainer Schuth <rainer.schuth@vnc.biz>
Dhaval Yoganandi <dhaval.yoganandi@vnc.biz>

June 5, 2015

Version 1.1

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1 VNCtalk Introduction

1.1 About Zimbra

Zimbra is a comprehensive communication and collaboration environment, fully called Zimbra Collaboration. Zimbra includes e-mail, contacts, calendars, tasks, instant messaging, file storage and web document management components. There are two editions of Zimbra available – the Open Source edition including all the functionality mentioned above, and the Network Edition which features full enterprise-class administration capabilities, plus mobile and desktop synchronization (of mail, contacts and calendar).

1.2 About VNCtalk

VNCtalk specializes in providing services for real time communication. It is a utility software which is installed on Zimbra as a Zimlet. The VNCtalk Zimlet enables its users to perform text chat, text conference, video chat and video conference and online document collaboration between users using Zimbra Zimlet technology.

1.3 Objectives and features

VNCtalk will help the user to interact and communicate more effectively with other users inside an enterprise network using the Zimbra UI without the need of installing extra software. Only a modern browser like *Chrome* is needed to start using VNCtalk. The communication across the users is always secure using TLS encryption.

Benefits

- Easy to use
- Encryption of user data
- Tight integration into the Zimbra environment
- Easy integration in corporate environments
- Single and Group Conversations
- Web-based Audio and Video conferencing using XMPP-jingle and WebRTC compatible technologies
- Platform independence due to the nature of XMPP
- No need of extra software on client-side
- Convert a text conference into a video conference
- Multiple independent video streams
- Integrated document sharing based on *Etherpad*

Features

- 1:1 text chat

- 1:n text chat
- 1:1 video chat
- 1:n video chat
- File transfer between 2 users
- Collaborative document creation (based on *Etherpad*)
- *LDAP* authentication for each user
- Automatic contact list population, based on Zimbra distribution lists

Features to come

- Screen sharing functionality
- File sharing *user2user* or *user2chatroom*
- External user login
- Mobile devices compatible

2 VNCtalk Architecture

2.1 System architecture Diagram

Figure 2.1 on page 4 show an overview of the VNCtalk architecture. Figure 2.2 on page 5 the relations between the components.

2.2 Components

VNCtalk consists of the interaction of several components running on different servers. These servers need to be reachable over the Internet or over an internal network, e.g. using a VPN.

1. Zimbra Zimlet
2. XMPP server
3. TURN/STUN server
4. Zimbra LDAP
5. VNCHybridAuth
6. Jappix mini
7. Jitsi
8. Etherpad
9. Screen sharing

The Zimbra Zimlet loads the mini chat client. It downloads the Javascript files directly from the XMPP server. Then the mini chat (based on the open source project Jappix Mini) connects to the XMPP server and provides the "1:n text chat (where n>=1)" We have enhanced the Jappix mini code to provide additional features like, additional multi-chat buttons and invite-person-to-chat buttons. Also a start video conference button was added. When a video conference is started, we use another component called Jitsi-Meet (which uses Jitsi-videobridge). Jitsi-Meet is the component which displays the video streams on the browser window and coordinates the call states (connect, stop, play, disconnect, and so on). The videobridge is like a network router but for video streams. Then we have a TURN/STUN server, which is used when clients are behind a NAT. This TURN server helps to forward video/audio traffic in complicate and not from the Internet accessible networks, like a NAT. Finally we use a hybrid authentication mechanism which also uses LDAP to authenticate the users. Based on Zimbra distribution lists, we build the user groups for the XMPP roster. The Etherpad component is used to provide document collaboration.

Zimbra Zimlet

This is a small Zimlet installed in the Zimbra machine which only takes care of loading the mini chat application (Jappix Mini) from the XMPP server. In the future this Zimlet will save some user settings to personalize the chat application during runtime.

Additionally there is the VNCtalk Admin Zimlet, which provides the persistent configuration like XMPP server domain and ports.

- XMPP server domain
- The domain where to get the Jappix Mini JS files

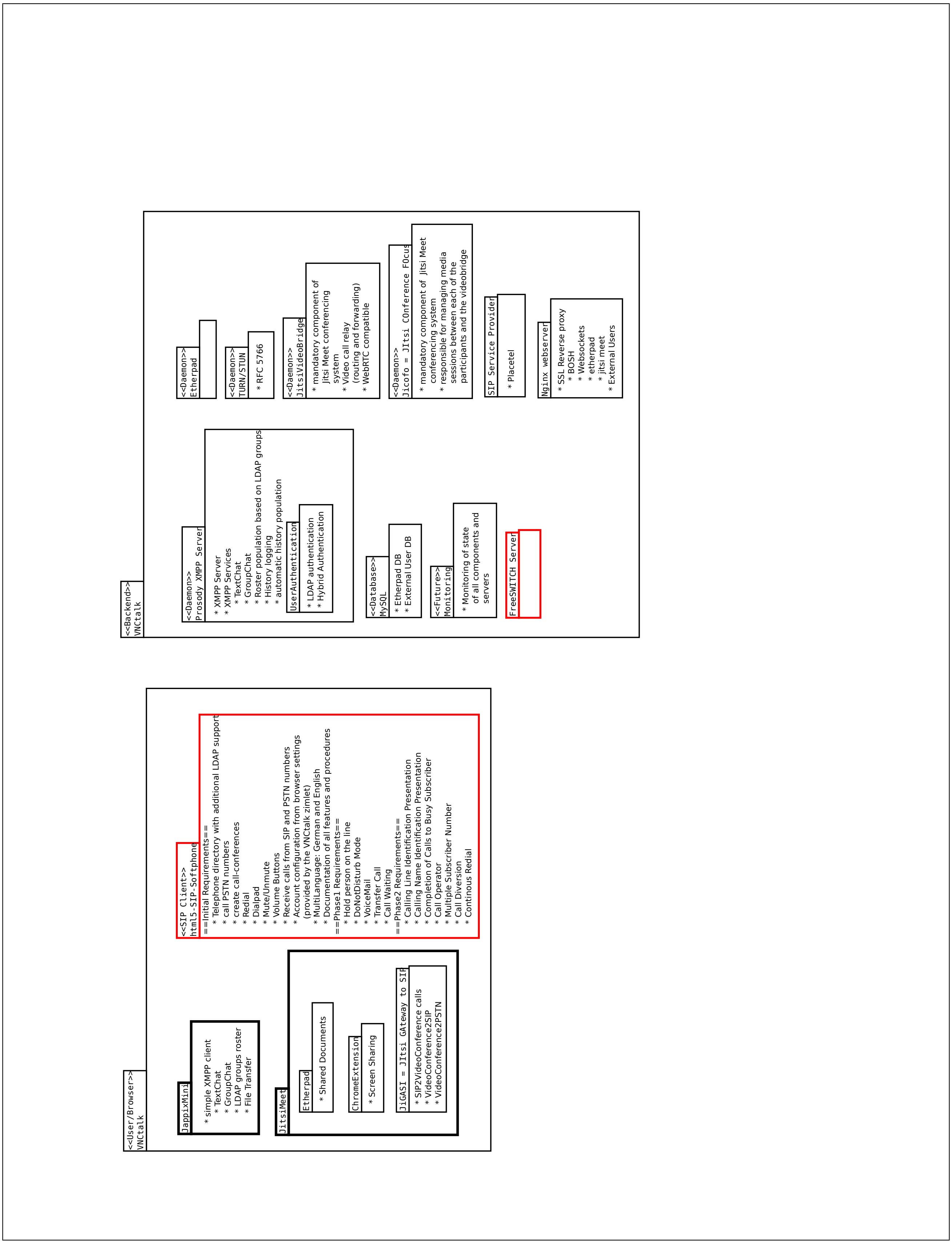

Figure 2.1: VNCtalk architecture overview

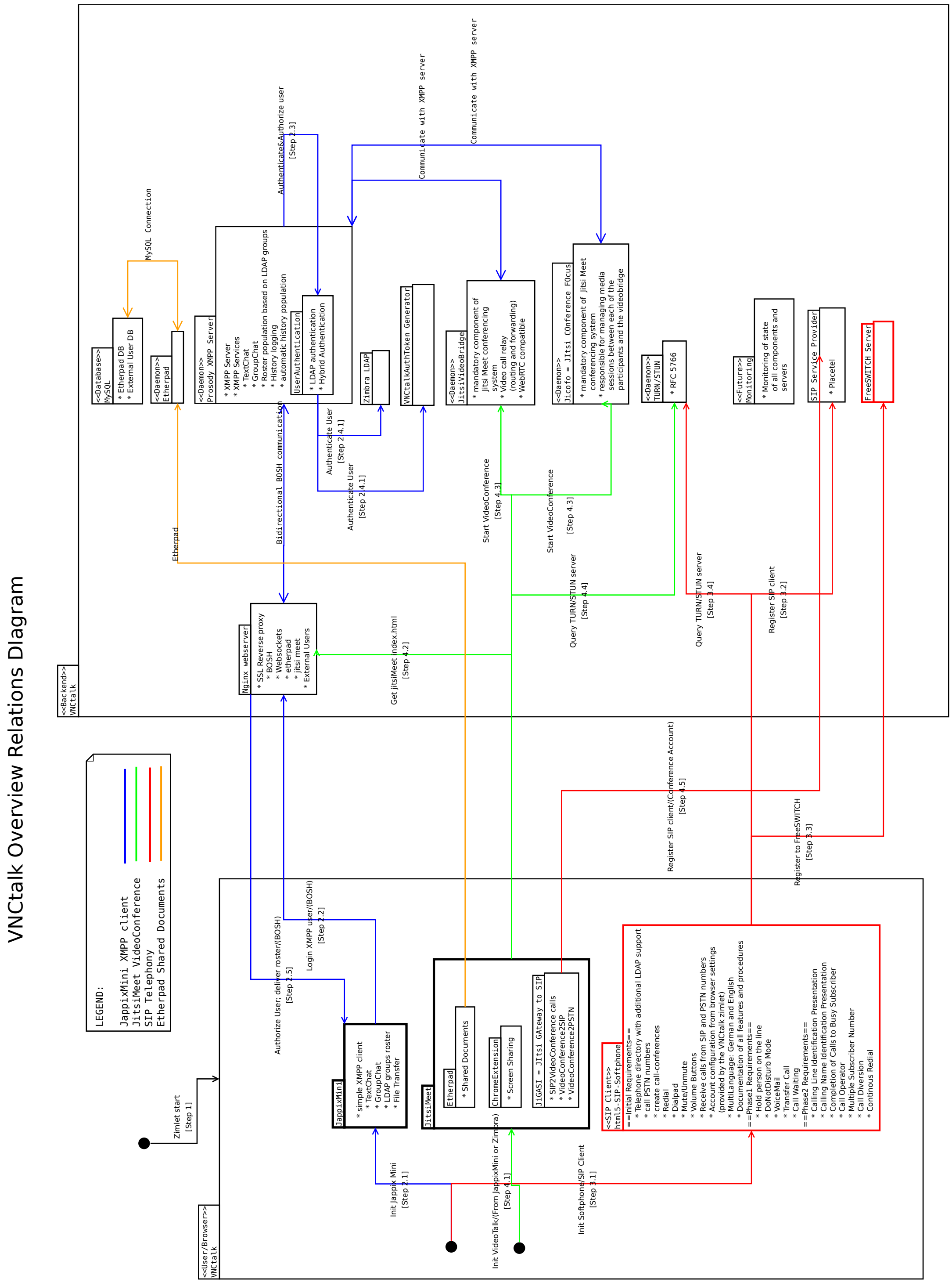

Figure 2.2: VNCtalk component relations

- Port numbers

XMPP server

We use the Open Source XMPP server *prosody*. Prosody is written in Lua and it is very easy to extend using modules. Prosody is used to manage the chat and multi-chat communication. In addition prosody is capable to authenticate users using LDAP and use an hybrid authenticate module written by VNC. The roster groups are also statically constructed using the Zimbra LDAP and distribution lists. Each time the distribution list changes, the *vnc_roster_-ldap* module must be restarted. Due to constraints of the XMPP protocol, the user needs to relogin to be able to get the changes in the roster.

TURN/STUN server

We use the *rfc5766-turn-server*.

LDAP

We use the LDAP daemon provided by Zimbra, but prosody could also use any other LDAP server. The usage of a different LDAP server is not discussed here.

VNCHybridAuth

To provide authentication of users based on our own criteria and against different user databases (LDAP, MySQL, etc.) a hybrid authentication module was written. This module is called by prosody asking him to authenticate the user by providing *user*,*host* and *password*. If the authentication was successful the hybrid authentication module exits with 0, otherwise returns 1.

Jappix mini

This is an Open Source project which provides a simple XMPP mini chat written in Javascript for websites. VNC extended this project with additional features and components and so be usable in the ZCS environment.

Jitsi

"Jitsi Meet is an OpenSource (MIT) WebRTC JavaScript application that uses Jitsi Videobridge to provide high quality, scalable video conferences." (https://jitsi.org/Projects/JitsiMeet)

Jitsi Meet also uses Jicofo (Jitsi Conference Focus). This is a mandatory component of Jitsi Meet conferencing system. It is responsible for managing media sessions between each of the participants and the videobridge.

Etherpad

Etherpad is a document collaboration software. With the help of Jitsi-meet Etherpad is used to provide to the users document sharing capabilities.

Screen sharing

This is provided by a Chrome extension.

2.3 DNS entries

This section lists the needed DNS entries for the XMPP and TURN server.

```
;;;
;;; VNC XMPP server
;;;
;;; A records for XMPP server
;;; OWNER-NAME                    TTL     CLASS   RR      IPV4
${XMPP_FULL_DOMAIN}.                  300     IN      A       ${PROSODY_IP4} ; Use if the server has this dns name
${PROSODY_FULL_HOSTNAME}.             300     IN      A       ${PROSODY_IP4}
prosody.${XMPP_FULL_DOMAIN}.          300     IN      A       ${PROSODY_IP4}
conference.${PROSODY_FULL_HOSTNAME}.  300     IN      A       ${PROSODY_IP4}
conference.${XMPP_FULL_DOMAIN}.       300     IN      A       ${PROSODY_IP4}
muc.${PROSODY_FULL_HOSTNAME}.         300     IN      A       ${PROSODY_IP4}
muc.${XMPP_FULL_DOMAIN}.              300     IN      A       ${PROSODY_IP4}
auth.${XMPP_FULL_DOMAIN}.             300     IN      A       ${PROSODY_IP4}
jitsi-videobridge.${XMPP_FULL_DOMAIN}.  300   IN      A       ${PROSODY_IP4}
focus.${XMPP_FULL_DOMAIN}.            300     IN      A       ${PROSODY_IP4}

;;;
;;; AAAA records for XMPP server
;;; OWNER-NAME                    TTL     CLASS   RR      IPV6
;;${XMPP_FULL_DOMAIN}.                  300     IN      A       ${PROSODY_IP6} ; Use if the server has this dns name
;;${PROSODY_FULL_HOSTNAME}.             300     IN      A       ${PROSODY_IP6}
;;prosody.${XMPP_FULL_DOMAIN}.          300     IN      A       ${PROSODY_IP6}
;;conference.${PROSODY_FULL_HOSTNAME}.  300     IN      A       ${PROSODY_IP6}
;;conference.${XMPP_FULL_DOMAIN}.       300     IN      A       ${PROSODY_IP6}
;;muc.${PROSODY_FULL_HOSTNAME}.         300     IN      A       ${PROSODY_IP6}
;;muc.${XMPP_FULL_DOMAIN}.              300     IN      A       ${PROSODY_IP6}
;;auth.${XMPP_FULL_DOMAIN}.             300     IN      A       ${PROSODY_IP6}
;;jitsi-videobridge.${XMPP_FULL_DOMAIN}.  300   IN      A       ${PROSODY_IP6}
;;focus.${XMPP_FULL_DOMAIN}.            300     IN      A       ${PROSODY_IP6}

;;;
;;; XMPP special records
;;;
;;; TXT records for BOSH and Websocket
;;; OWNER-NAM                           TTL CLASS RR  TEXT
_xmppconnect.${PROSODY_FULL_HOSTNAME}. 300 IN    TXT "_xmpp-client-xbosh=https://${PROSODY_FULL_HOSTNAME}:443/http-bind"
_xmppconnect.${XMPP_FULL_DOMAIN}.      300 IN    TXT "_xmpp-client-xbosh=https://${PROSODY_FULL_HOSTNAME}:443/http-bind"

;;;
;;; SRV records for XMPP
;;; SRVCE.PROT.OWNER-NAME                         TTL     CLASS   RR  PRI     WEIGHT  PORT    TARGET
_xmpp-client._tcp.${XMPP_FULL_DOMAIN}.                 300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.${XMPP_FULL_DOMAIN}.                 300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.prosody.${XMPP_FULL_DOMAIN}.         300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.prosody.${XMPP_FULL_DOMAIN}.         300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.auth.${XMPP_FULL_DOMAIN}.            300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.auth.${XMPP_FULL_DOMAIN}.            300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.focus.${XMPP_FULL_DOMAIN}.           300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.focus.${XMPP_FULL_DOMAIN}.           300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.jitsi-videobridge.${XMPP_FULL_DOMAIN}. 300   IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.jitsi-videobridge.${XMPP_FULL_DOMAIN}. 300   IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.conference.${XMPP_FULL_DOMAIN}.      300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.conference.${XMPP_FULL_DOMAIN}.      300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.conference.${PROSODY_FULL_HOSTNAME}. 300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.conference.${PROSODY_FULL_HOSTNAME}. 300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.muc.${XMPP_FULL_DOMAIN}.             300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.muc.${XMPP_FULL_DOMAIN}.             300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.
_xmpp-client._tcp.muc.${PROSODY_FULL_HOSTNAME}.        300     IN      SRV 0       5       5222    ${PROSODY_FULL_HOSTNAME}.
_xmpp-server._tcp.muc.${PROSODY_FULL_HOSTNAME}.        300     IN      SRV 0       5       5269    ${PROSODY_FULL_HOSTNAME}.

;;;
;;; SRV records for STUN/TURN
;;; SRVCE.PROT.OWNER-NAME     TTL     CLASS   RR  PRI     WEIGHT  PORT    TARGET
;;
;;THIS RECORDS HAVE NOT BEEN TESTED. Only for FUTURE use.
;;;
;;_stun._tcp.${XMPP_FULL_DOMAIN}.       300     IN      SRV 0       5       3478    ${PROSODY_FULL_HOSTNAME}.
;;_stun._udp.${XMPP_FULL_DOMAIN}.       300     IN      SRV 0       5       3478    ${PROSODY_FULL_HOSTNAME}.
;;_stuns._tcp.${XMPP_FULL_DOMAIN}.      300     IN      SRV 0       5       5349    ${PROSODY_FULL_HOSTNAME}.
```

```
;;_turn._tcp.${XMPP_FULL_DOMAIN}.        300   IN      SRV 0       5       3478    ${PROSODY_FULL_HOSTNAME}.
;;_turn._udp.${XMPP_FULL_DOMAIN}.        300   IN      SRV 0       5       3478    ${PROSODY_FULL_HOSTNAME}.
;;_turns._tcp.${XMPP_FULL_DOMAIN}.       300   IN      SRV 0       5       5349    ${PROSODY_FULL_HOSTNAME}.
```

2.4 Supported Zimbra version

Zimbra 8.0 (Iron Maiden) and 8.6 (Judas Priest) are officially supported. Zimbra 7.0 (Helix) is not officially supported.

2.5 Limitations

The usage of the Chrome or Chromium browser is currently mandatory. Other browsers like Firefox, Opera or Safari are not supported and VNCtalk will not run on those browsers!

A functional webcam and microphone&headphones are also needed for videochats.

The available bandwidth to the user is also a limitation. If many users, e.g. more than 6 users, are using the videochat application at the same time, the complete uplink and downlink bandwidth of the Internet could be saturated, possibly affecting the quality of the conversation.

3 SIP integration into VNCtalk

3.1 The Purpose of the Project

We need an independent Softphone/sip-client component to integrate in the browser inside VNCtalk. This component should be easy to integrate by calling the needed Javascript libraries and scripts. The Softphone will register itself as a stand-alone SIP client and should be able to replace a phone device like the Snom 320 (`https://www.snom.com/en/products/snom-efficient-line/snom-320`).

Image 3.1 shows the Snom 370 telephone, capable of SIP telephony.

Figure 3.1: Snom370 telephone

All features and procedures must be documented in a PDF file in English language. This documentation must contain the following:

1. Description of each feature in the Softphone application and its API call.

2. What does each feature, e.g. The green button on the dialpad starts a SIP call using the number entered by the user.
3. Provide examples on how to perform the API call.
4. The provided code must be documented in doxygen[1] format, so an automated documentation can be made.

Goals of the Project

- The Softphone will register itself as a SIP client and should be enable to replace a phone device like the Snom 320.
- It should be written in HTML5 and Javascript, so it can be used on most browsers and mobile platforms.
- Enable the user to communicate using voice communication with other parties using SIP. (Make and receive calls).
- Enable the user to query and select the number to be dialed from a telephone directory provided by the Zimbra LDAP.
- Enable the user to enter or create telephone conferences.
- Enable the user to access a running VNCtalk videoconference using SIP and Jigasi.
- Enable the user to communicate in complex network environments: use STUN/TURN if needed.

3.2 User Interface

The Softphone user interface should look like a normal SIP capable telephone, like the Snom 320 (`https://www.snom.com/en/products/snom-efficient-line/snom-320`).

The interface should be similar to interface showed in Image 3.2. This interface present the user with a full dial-pad, volume and mute buttons. It also gives the user to access certain features directly, like contact directory or conference call creation/join. Additionally the interface must also have a “display” to present information to the user, see Image 3.1. The Softphone must not implement all the features and buttons showed in Image 3.2, the Softphone must only implement the features required by official requirements list presented in section

[1] Other documentation format can be used, if needed.

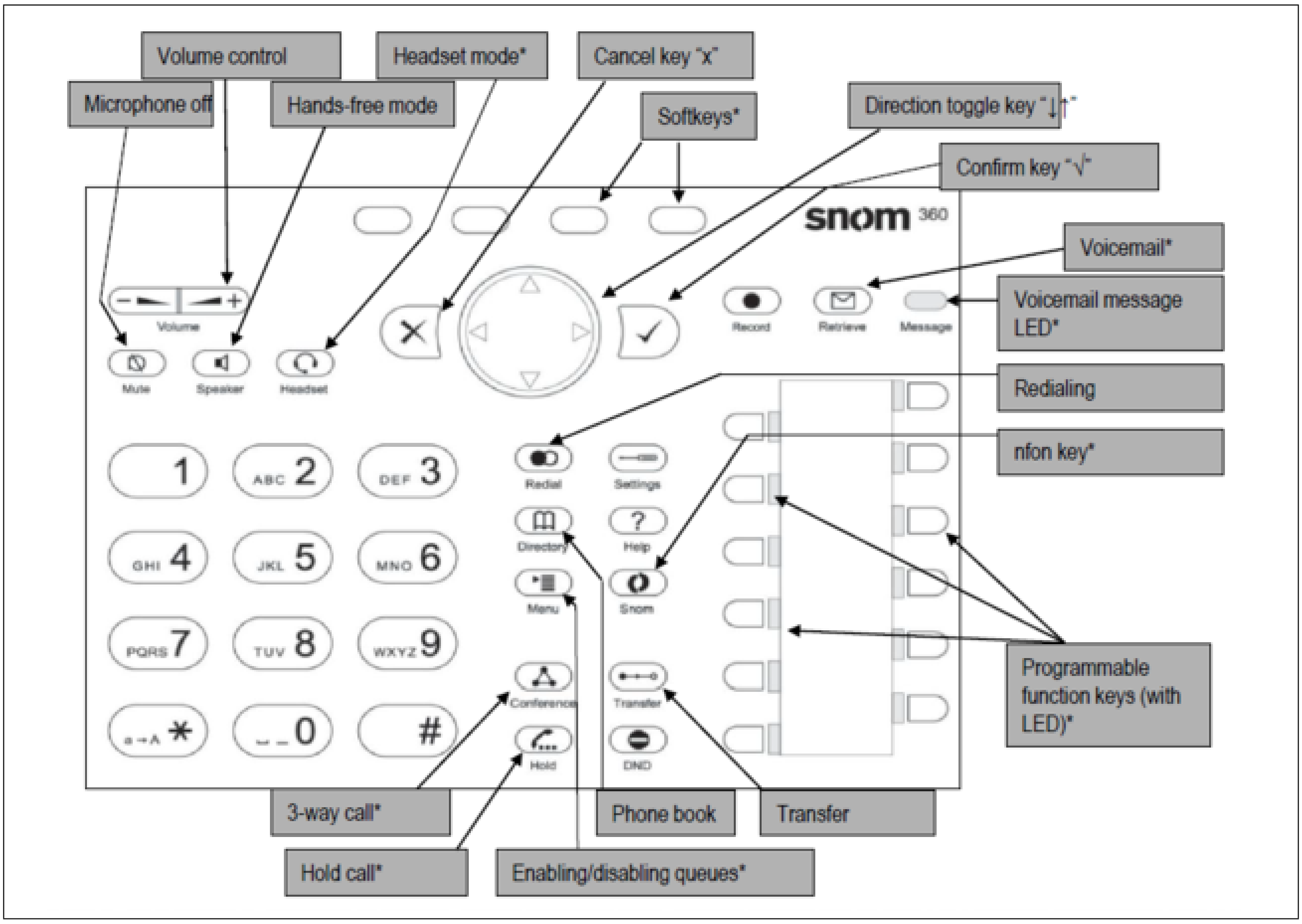

Figure 3.2: Snom keys layout

3.3 Use Case Scenarios

Figure 3.3 on page 12 described graphically the call flows of scenario 1 & 2.

Scenario 1

1. adding user to existing (video-)conference using API calls.
2. the invited user will get a phone call
3. answering the call he will be in the conference

Scenario 2

1. Using the Softphone, a user, who is not in the (video-)conference, will call the conference SIP number
2. and then join the ongoing (video-)conference.

Scenario 3

1. An (external) PSTN user calls the SIP number of the conference

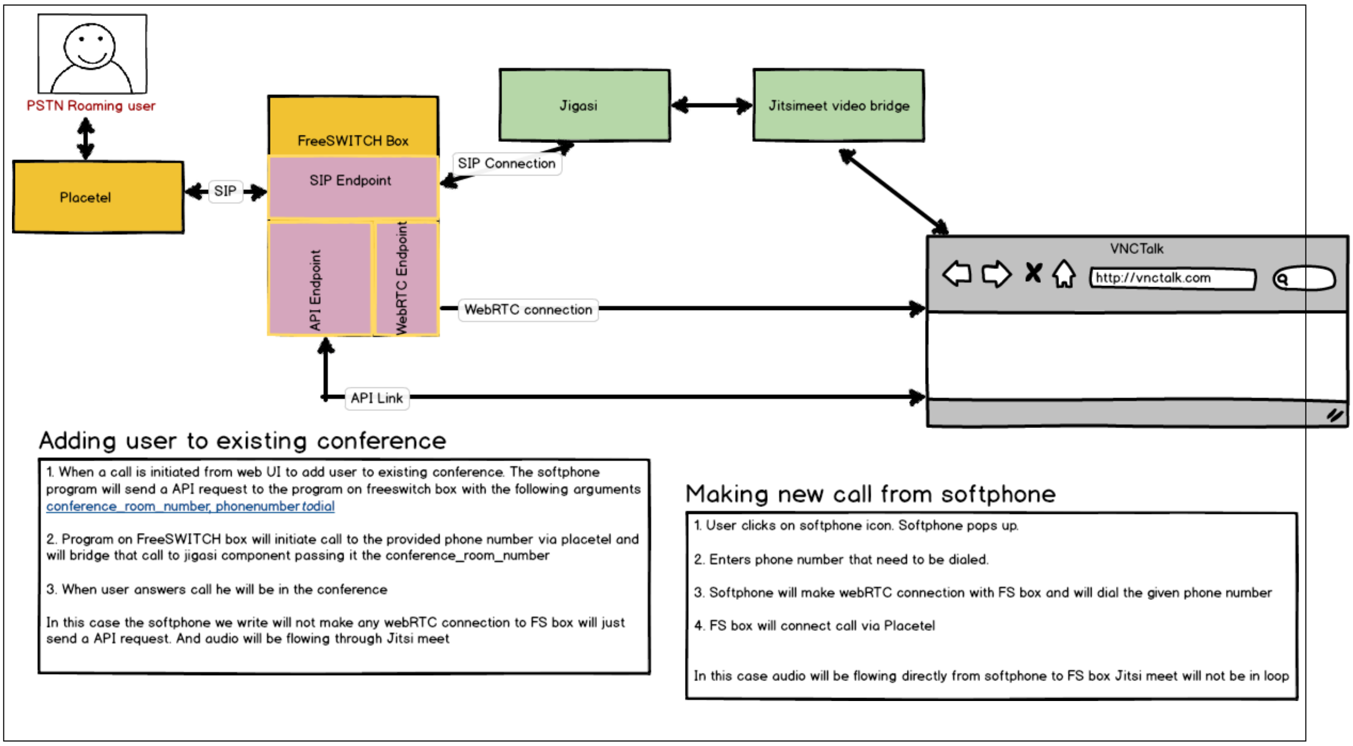

Figure 3.3: Call flow Scenario

2. and is able to join the ongoing (video-)conference.

3.4 Project Constraints

Mandated Constraints

- HTML5, Javascript based Softphone/SIP client
- Multi language support: German and English
- Zimbra LDAP integration
- Integration with Jigasi https://github.com/jitsi/jigasi.

Implementation Environment of the Current System

The environment is the Zimbra collaboration suite in the versions 8.0 (iron maiden) and 8.6 (judas priest). The application must run inside this environment as an external component called by the VNCtalk Zimbra Zimlet. The VNCtalk zimlet will provide the needed credentials to the Softphone/SIP client during initialization.

Schedule Constraint

The initial features must be finished and tested after 30 days or starting the development.

Budget Constraints

FIXME.

3.5 Naming Conventions and Definitions

Activity is a procedure performed by the Softphone or by the user.
For example: *start a call* is an activity. *Join a conference* is another activity.

Application the Softphone/SIP client

SIP denotes the Session Initiation Protocol but also the other needed media stream protocols (RTP, WebRTC).

SIP number/address denote a SIP account on a SIP server, e.g. 012345@sip.net, or rainer.schuth@sip.net .

User the person who uses the application during normal work.

Zimbra the Zimbra collaboration environment on which the application must run.

3.6 Phase 1: Initial Requirements

After finishing the initial phase, we will then decide if we go with Phase 2 and Phase 3.

The order and number of the requirement does not reflect the importance of the requirement! All requirements listed here are required!

- **Requirement Number 0**: SIP sever independent Softphone
  Description: The Softphone must be able to connect to any SIP server (or similar server) using standardized SIP technologies. The Softphone must be able to login as a normal SIP-capable telephone using SIP credentials (username, register, password, port).
  Rationale: independent Softphone inside the browser.
  Fit Criterion: The Softphone must be able to log in with any SIP server using standardized SIP login methods.
- **Requirement Number 1**: Call PSTN numbers
  Description: The user should be able to dial a PSTN number using the dialpad or a phone number provided by the telephone directory.
  Rationale: User wants to call "real" PSTN numbers.
  Fit Criterion: The application provides the capability to dial a PSTN numbers using SIP.
- **Requirement Number 2**: Call SIP numbers
  Description: The user should be able to dial a SIP number/address using the dialpad or a sip address provided by the telephone directory.
  Rationale: User wants to call SIP addresses.
  Fit Criterion: The application provides the capability to enter a SIP address and call it using SIP.
- **Requirement Number 3:** Receive call from SIP and PSTN numbers
  Description: The user should be able receive a call from any PSTN and SIP number which is trying to contact the user's SIP address.
  Rationale: User wants to receive telephone calls to the application.
  Fit Criterion: The application provides the capability to receive calls using SIP.

- **Requirement Number 4:** dialpad
  Description: The application provides a dialpad so the user is able to enter numbers to be dialed using SIP.
  Rationale: User wants to enter numbers which are not listed in the telephone directory.
  Fit Criterion: The application provides the capability to enter telephone and SIP numbers to be dialed.
- **Requirement Number 5:** Mute/unmute buttons
  Description: The application provides a buttons to suppress/enable the user's microphone, so the audio signal is not transmitted if the mute button is active.
  Rationale: User wants to mute his microphone during some calls due to different reasons.
  Fit Criterion: The application provides the capability mute/unmute the user's microphone.
- **Requirement Number 6:** Volume buttons
  Description: The application provides a buttons to increase or decrease the volume of the receiving audio signal.
  Rationale: User wants to control the volume of the receiving audio call.
  Fit Criterion: The application provides the capability to increase or decrease the volume of a call.
- **Requirement Number 7:** Multi language support
  Description: The application must provide a mechanism to enable multi language support, based on the browser locale or the language definition provided by the Zimbra environment. Initially the application must provide support fro the German and English language.
  Rationale: User does not always speaks English, therefore German support is needed.
  Fit Criterion: The application provides the capability change the language based on the users locales.
- **Requirement Number 8:** Documentation of the APIs
  Description: The application must provide well documented APIs to perform the needed actions, like start call, drop call; basically every action performed by the Softphone should be callable by a defined API.
  Rationale: VNC developers want to access the APIs provided by the application, e.g. to start a telephone call.
  Fit Criterion: Documentation of the APIs is needed
- **Requirement Number 9:** API calls to start different activities
  Description: The application provides a set of APIs which can be used by other applications running on the browser session to start any activity supported by the application.
  Rationale: The VNCtalk Zimlet wants to perform an activity provided by the Softphone, e.g. start a call by selecting a user from the XMPP roster list and clicking on the *call* button for the selected user.
  Fit Criterion: The application provides APIs to perform SIP/telephony related activities.
- **Requirement Number 10:** Telephony directory with additional LDAP support
  Description: The application provides a mechanism to access and modify the telephony directory provided by the Zimbra LDAP. In this directory telephone numbers and SIP numbers are saved with the respective descriptions, e.g +4913344455 is the phone number of the VNC offices.
  Rationale: User wants to access a telephone directory, where he saves his telephone numbers and SIP numbers.
  Fit Criterion: The application provides the capability to access and modify the LDAP directory.
- **Requirement Number 11:** Create telephone conferences
  Description: The application provides a mechanism to create a telephone conference to which other users can access by calling the conference number. The conference service will be provided by an external service provider.
  Rationale: User wants to create a conference call to discuss with colleges about a topic.
  Fit Criterion: The application provides the capability to create a telephone conference.
- **Requirement Number 12:** Join telephone conferences
  Description: The application provides a mechanism to join a telephone conference. The conference service will be provided by an external service provider.
  Rationale: User wants to join a conference call to discuss with colleges about a topic.
  Fit Criterion: The application provides the capability to join a telephone conference.

- **Requirement Number 13:** Join JitsiMeet video conference over SIP using Jigasi
Description: The application provides a mechanism to join a running JitsiMeet video conference using SIP and Jigasi.
Rationale: User wants to join a running JitsiMeet conference to discuss with colleges about a topic, but he does not want or can use the video chat, because he is driving a car.
Fit Criterion: The application provides the capability to join a JitsiMeet conference.
- **Requirement Number 14:** Redial button
Description: The application provides button to enable to redial the last called number.
Rationale: User wants to call the last number again without the need to enter the phone number/SIP account again.
Fit Criterion: The application provides the capability to redial the last called (SIP) number.
- **Requirement Number 15:** Account/Credentials configuration
Description: The VNCtalk Zimbra environment will provide the needed credentials to log in to the SIP servers. The application must be able to use the credentials provided by VNCtalk. A global variable will contain the needed login data.
Rationale: Configure the application externally during initialization.
Fit Criterion: The application can use the provided settings.

3.7 Phase 2: Further Requirements

- **Requirement Number 17:** Hold person on the line
Description: Through the feature hold a subscriber can bring an existing connection to a wait state. The other party gets into this state usually an announcement or music on hold.
- **Requirement Number 18:** DoNotDisturb Mode
Description: Using this feature the user who activates this, will be not disturbed by incoming calls, because this calls not ring. Instead these calls will be redirected directly to the mailbox. In addition the user must be informed that this feature is active by coloring the phone red.
- **Requirement Number 19:** Voice Mail
Description: The application provides a mechanism to access the voice mail.
- **Requirement Number 20:** Transfer Call
Description: The application provides a mechanism to transfer an incoming or ongoing call to another telephone/SIP numbers.
- **Requirement Number 21:** Call Waiting
Description: If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the called party is able to suspend the current telephone call and switch to the new incoming call (typically, this is done by pushing the flash button), and can then negotiate with the new or the current caller an appropriate time to ring back.

3.8 Phase 3

- **Requirement Number 22:** Calling Line Identification Presentation
- **Requirement Number 23:** Calling Name Identification Presentation
- **Requirement Number 24:** Completion of Call of Busy Subscriber

- **Requirement Number 25:** Call operator
- **Requirement Number 26:** Multiple Subscriber Number
- **Requirement Number 27:** Call Diversion
- **Requirement Number 28:** Continuous Redial

4 Developing environment

VNC will provide a developing environment accessed using VPN, from which developers can test the integration with VNCtalk and Zimbra.

VNC
Virtual Network Consult

About VNC

VNC - Virtual Network Consult is a leading business cloud integrator and a specialist in commercial open source solutions. VNC provides services to organizations and businesses of all sizes - from small and medium businesses to globally operating corporations. Our services range from platform data center solutions to complex business applications such as the VNCportal, CRM-ERP, VNCmail+ and advanced Secure Communications Environments – mobile and suited to the needs of each client.

VNC Contact

pr@vnc.biz ; www.vnc.biz

VNC - Virtual Network Consult AG

Poststr. 24, CH - 6301 Zug, Switzerland

Phone: +41 (41) 727 52 00

VNC - Virtual Network Consult GmbH

Pariser Platz 4a, D-10117 Berlin, Germany

Phone: +49 (30) 3464615-20