**#18536** - <https://redmine.vnc.biz/issues/18536>

1. **Description**:

Setting up Jigasi

1. **Analytic**
   1. **Root cause**:
   2. **Solutions**:

* I follow this guide to install jigasi in prosody-dev4: <https://redmine.vnc.biz/projects/vnc-uc/wiki/VNCTalkSoftphoneJiGaSi>
* I config Jigasi use SIP user 1007@sip-dev1.vnctalk.zimbra-vnc.de to connect to sip-dev1 server. Config jigasi  
  JIGASI\_SECRET=vnctalk  
  JVB\_HOSTNAME=prosody-dev4.vnctalk.zimbra-vnc  
  Config jigasi use username and password of prosody-dev4 server
* I can start prosody server with component callcontrol.prosody-dev4.vnctalk.zimbra-vnc.de
* I got error "Server-to-server connection failed: DNS resolution failed". I see "connection.emuc.focusMucJid" is null when I make outgoing call .   
  I work around and try some config from <https://github.com/jitsi/jigasi> but still get the same error.
* I try hardcore focusMucJid to "*xxx@conference.prosody-dev4.vnctalk.zimbra-vnc.de/focus*". Stanza iq result from prosody server is OK

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| *<iq xmlns="jabber:client" type="result"*  *to="alice@prosody-dev4.vnctalk.zimbra-vnc.de/906a9b84-4bb3-4e26-849b-76fe8215d7c4"*  *from="xxx@conference.prosody-dev4.vnctalk.zimbra-vnc.de/focus" id="9:sendIQ">*  *<ref xmlns="urn:xmpp:rayo:1"uri="xmpp:14fac6131d7@callcontrol.prosody-dev4.vnctalk.zimbra-vnc.de"/></iq>* |

But sip-dev1 does not receive any SIP message.

* The connection between jigasi - sip-dev1 is fine, I can see user 1007 registered in sip-dev1. I think the problem come from the jitsi-meet source because we install jicofo, jitsi-video-bridge and jigasi from ubuntu package but install jitsi-meet from github source. I will try to install jitsi- meet from package in my local server to verify that.
* I have installed jitsi-meet, jicofo, jitsi-videobridge, jigasi and Freeswitch from ubuntu packages in my local server. I can make outbound call from video conference to softphone, voice is okay.
* I will check the version of jitsi-meet package and compare with our current version to find out root cause
* When I tried to install the jigasi from package, found the problem: it also require install jitsi-meet and jitsi-prosody packages but we use jitsi-meet from source:

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| **dpkg -i jigasi\_1.0-81\_amd64.deb**  *Selecting previously unselected package jigasi. dpkg: regarding jigasi\_1.0-81\_amd64.deb containing jigasi, pre-dependency problem:*  *jigasi pre-depends on jitsi-meet*  *jitsi-meet is not installed but configs remain. dpkg: error processing archive jigasi\_1.0-81\_amd64.deb (--install): pre-dependency problem - not installing jigasi Errors were encountered while processing: jigasi\_1.0-81\_amd64.deb* |

Then I change to set up jigasi from source:

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| git clone https://github.com/jitsi/jigasi cd jigasi git checkout b231ff38da7135878fef89dd90eb99e871c3e779 (release 84 is same version with package install) ant make |

* Set up local server with the same jitsi-meet on release 2.5.0 as Quy Dang told me:

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| [*https://github.com/jitsi/jitsi-meet/commit/239f271caf81080fc34328e22ea99cc7351079eb*](https://github.com/jitsi/jitsi-meet/commit/239f271caf81080fc34328e22ea99cc7351079eb)  *Jicofo: 1.0-109-1*  *Jitsi-videobrigde: 472-1*  *Following this ticket:* [*https://redmine.vnc.biz/issues/19868*](https://redmine.vnc.biz/issues/19868)  *All jitsi-packages are same version with current prosody* |

* I can make the outgoing/incoming call with prosody-authentication mode="anonymous".
* Now I will set up jigasi from source on prosody-dev4.
* I have set up jigasi from source on prosody-dev4 with jitsi-meet on release 2.5.0. And I have modified code in libs/app.bundle.js and modules/xmpp/strophe.emuc.js file:

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| *if(from.indexOf('\_video') == -1 && from.indexOf('focus') == -1)  return true;* |

to update connection.emuc.focusMucJid. Then we can pass to "Server-to-server connection failed: DNS resolution failed".

* I also work around to pass external user for jigasi@prosody-dev4.vnctalk.zimbra-vnc.de user. I have test incoming/outgoing call between user in siptest@conference.prosody-dev4.vnctalk.zimbra-vnc.de and 1008@10.20.20.51. The signal worked fine but the audio was only one-way. Only 1008@10.20.20.51 can get the media on both incoming/outgoing call. I also test with sip2sip.info and it's still there.
* I guess the problem may be the NAT on Firewall, but need to debug more. I have already apply port forwarding on FireWall for prosody-dev4

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| Port: 26001-26500 Map-Port: 26001-26500  IP: 10.20.20.44 Map-IP: 193.254.187.146  Protocol: UDP |

* I have test the case:
* User Alice create conference room siptest, then, Alice invite Bob to join room --> It work fine and have audio two way.
* After that, Alice call sip user 1008@10.20.20.51 --> It work fine and have audio on all client.
* Alice disconnect all member in the room (Bob and 1008). Then, call sip user 1008@10.20.20.51 --> Have audio two way. Also test which sip2sip.info and it work too. Incoming and outgoing call.

==> So I guess the root cause: the user at the first time when join video conference alone and call SIP account via Jigasi can't get audio stream.

* After compare the current code of vnctalk-jitsi-meet and original source code of Github (same commit). I found the reason is media stream can not play because of missing the small video at bottom-right screen. So, it can not be play remoteAudio stream.
* I see the resource of user Jigasi have been created random string, so I enable in jigasi/jigasi-home/sip-communicator.properties file:

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| *org.jitsi.jigasi.USE\_SIP\_USER\_AS\_XMPP\_RESOURCE=true* |

to prevent jigasi create random resource each session.

* I have hard code in libs/app.bundle.js and modules/xmpp/strophe.emuc.js file for test call will sip user quydang@sip2sip.info:

The jigasi will use the resouce: jigasi@prosody-dev4.vnctalk.zimbra-vnc.de/quydang, so we can filter:

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| if(from.indexOf('\_video') == -1 && from.indexOf('focus') == -1 && from.indexOf('quydang') == -1)  return true; |

==> I can call with sip user quydang@sip2sip.info and have audio in two ways with small video of sip user showed.

1. **Implementation**
   1. **Code:**