**#22086** - <https://redmine.vnc.biz/issues/22086>

1. **Description**:

Put the sipml5 sofphone logic inside the vnctalk softphone.

Use the official sipml5 sources as an example: <https://redmine.vnc.biz/projects/vnctalk-sipml5>

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

* Apply the SIPml5 softphone logic inside VNCtalk softphone:

+ Include softphone/js/sipml5/SIPml-api.js

+ Include expert setting page and css.

+ Create call.js follow the logic of file <https://redmine.vnc.biz/projects/vnctalk-sipml5/repository/changes/release/call.htm?rev=master>

+ Hard code user registration:

softphone/js/call.js

|  |
| --- |
| / sends SIP REGISTER request to login function sipRegister() {  txtDisplayName = "quydang";  txtPrivateIdentity = "quydang";  txtPublicIdentity = "sip:quydang@sip2sip.info";  txtPassword = "12345678@X";  txtRealm = "sip2sip.info"; ... |

* Edit index.html and sofphone.js to work with call.js.
* Test some cases with sip2sip.info provider:

+ Out going call

+ In coming call

+ Mute/unmute button

+ Hang up

+ Display status call on monitor - reference SIPml5.

* How can we test it ?

1) Set up nginx server and configure the source path to softphone folder.

2) Create new user on sip2sip.info, <https://mdns.sipthor.net/sip_settings.phtml>

Please login to another softphone like Zoiper,...

3) Need to configure mode expert setting at <your\_webserver\_IP>/expert.htm

|  |
| --- |
| Disable Video: [X] Enable RTCWeb Breaker[1]: [X]  WebSocket Server URL[2]: ws://10.20.20.26:15090 SIP outbound Proxy URL[3]: udp://proxy.sipthor.net ICE Servers[4]: [{ url: 'stun:stun.l.google.com:19302'}] Max bandwidth (kbps)[5]: Video size[6]:  Disable 3GPP Early IMS[7]:  Disable debug messages[8]:  Cache the media stream[9]: [X] Disable Call button options[10]: |

5) You can start testing.

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