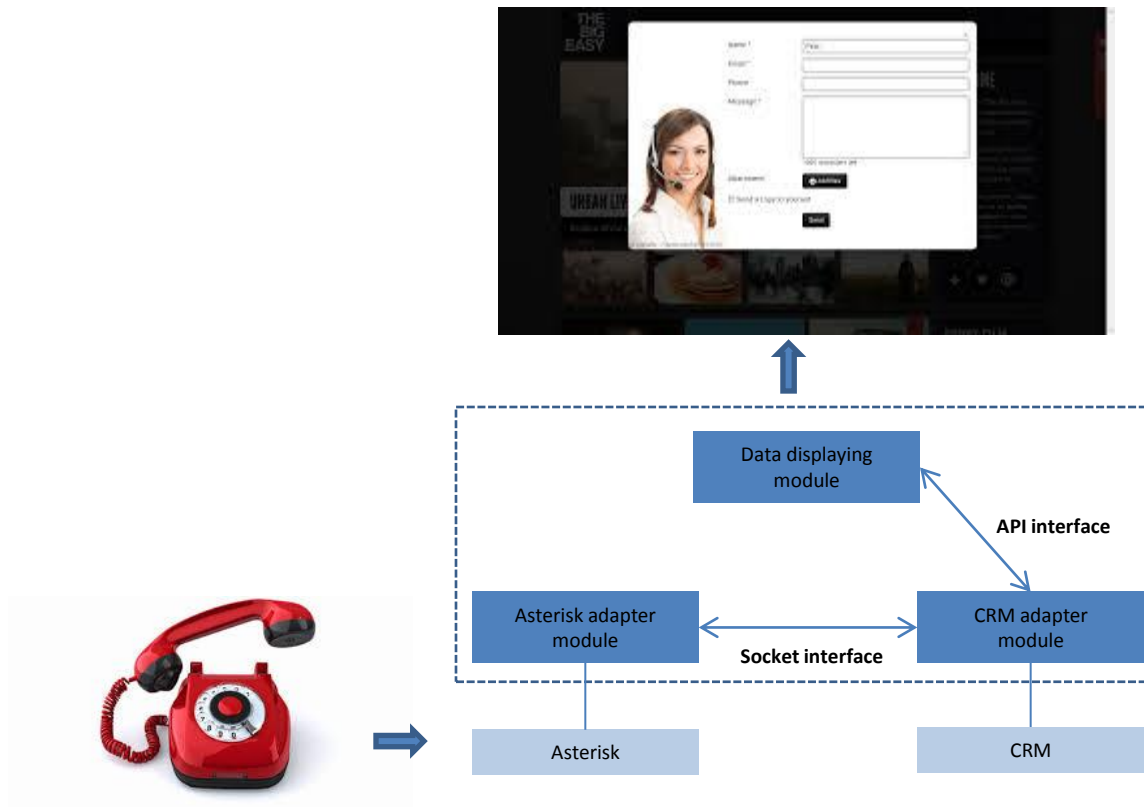


General diagram



SOME REMARKS

1) Asterisk adapter module

This module should run on server side, the machine at which Asterisk is deployed.

2) CRM adapter module

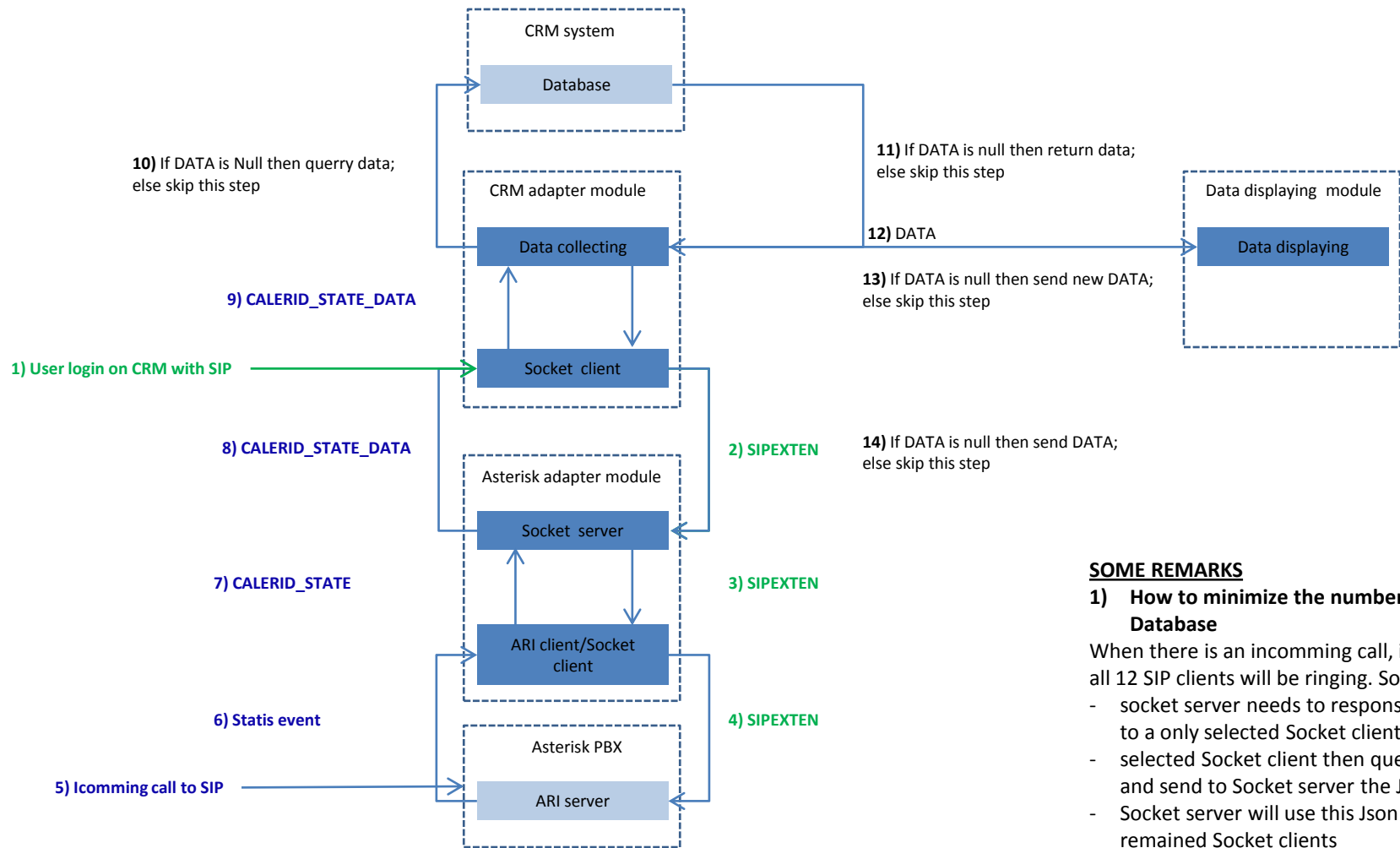
This module run on browser

3) Data displaying module

This module run on browser

4) Number of max calls is 12

Flow to retrieve customer info by by CallerID of a SIP client



SOME REMARKS

1) How to minimize the number of queries to Database

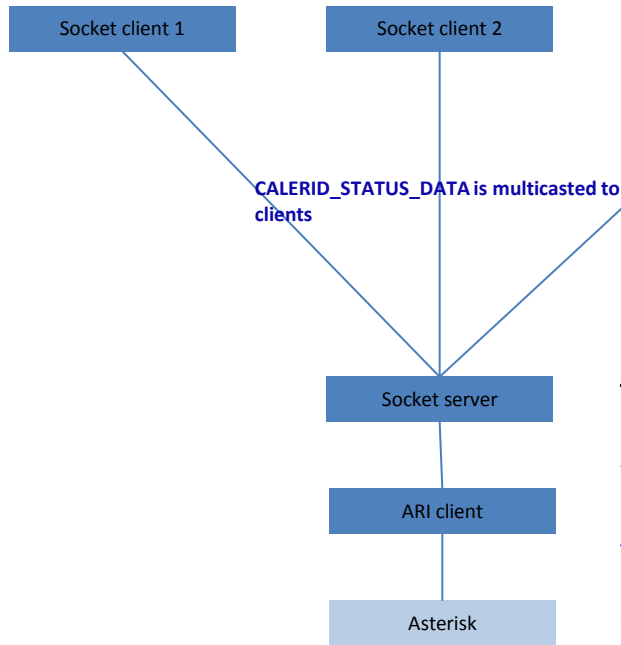
When there is an incoming call, in case of Onepas, all 12 SIP clients will be ringing. So :

- socket server needs to response CallerID & status to a only selected Socket client
- selected Socket client then queries data from DB and send to Socket server the Json data
- Socket server will use this Json data to response 11 remained Socket clients

2) How to choose a selected Socket client

- Preselect
- Or randomly

How to minimize the number of connections to Asterisk and the number of Stasis apps as well



SOME REMARKS

1) 12 socket clients run on 12 Web Browsers

Each socket client is identified by its SIP exten number; When a client connects, server adds its SIP exten to SIP_EXTEN_LIST. When that client disconnects, its SIP exten is removed.

Each SIP softphone has a SIP outbound channel, when receiving incoming calls, according to <https://wiki.asterisk.org/wiki/display/AST/Channels>

2) How to minimize the number of callings to Asterisk ARI

When socket server starts, it creates 01 ARI client and 01 Stasis application

3) Socket server creates only 1 ARI client

Server should maintain a Outbound channel status tables

The Outbound channel status could be collected by wscat -c

ws://localhost:8088/ari/channels?api_key=asterisk:asterisk

Status table:

[SIP extension][Outbound channel id][Outbound channel status]

Channel status are included in <https://www.voip-info.org/wiki/view/channel+status>:

- 2 Channel is off hook
- 4 Line is ringing

Server regularly multicast CALLERID_STATUS_DATA to connected clients

4) On Asterisk, there is only 1 running Stasis application for all SIP extensions

Stasis app will return the information of the Inbound channel, which is clarified at

<https://wiki.asterisk.org/wiki/display/AST/Channels>

CALERID_STATE_DATA

Each incoming call will have a CALLERID_STATE_DATA

Extens in CALLERID_STATE_DATA must be existed in SIP_EXTEN_LIST

```
{ "callerId": "84904859696",
  "state": [
    { "exten": "1000", "outboundChannelId": "1234567.89", "outboundChannelState": "Ringing",
      ...
    { "exten": "2000", "outboundChannelId": "7654321.98", "outboundChannelState": "Up",,,
  ],
  "data": {
    "phone": "+84904958686"
    "name": "Nguyễn Văn A",
    "lastBooking": { "info": "Ngày 10/08/2017, 2 người lớn, 2 trẻ em, tại nhà hàng Song Dương Văn Cao", "state": "Chờ xác nhận"},
    "totalBookingTimes": { "info": "20", "linkDetails": "https://partner.onepas.vn/abdefgh"},
    "totalCancelingTimes": { "info": "2" },
  },
  "outboundChannelId": "9999999.999",
  "duration": { "start": "2017/09/18 06:12:42", "end": "2017/09/18 06:12:42"},
}
```

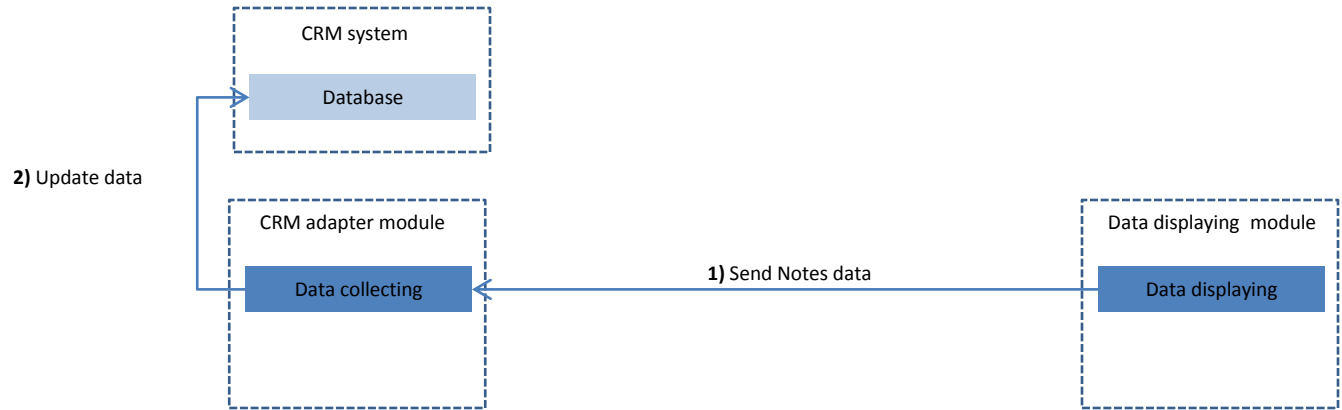
SIP_EXTEN_LIST

All socket clients, which have extens in SIP_EXTEN_LIST, must be connected

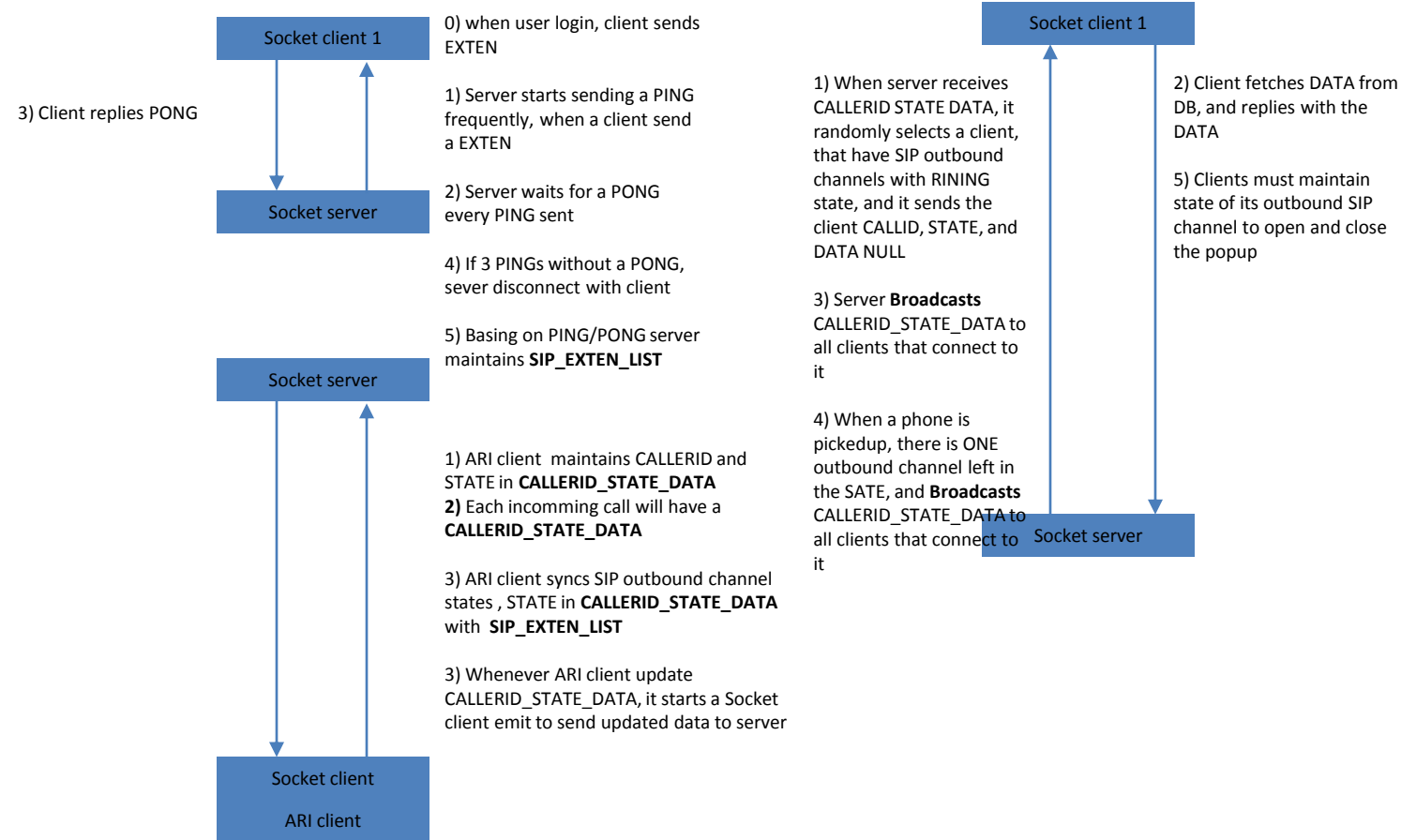
to socket server

```
[
  { 'exten': 1000, 'pingCounter': 0, 'socket': socket1 },
  { 'exten': 2000, 'pingCounter': 0, 'socket': socket2 },
  ...
]
```

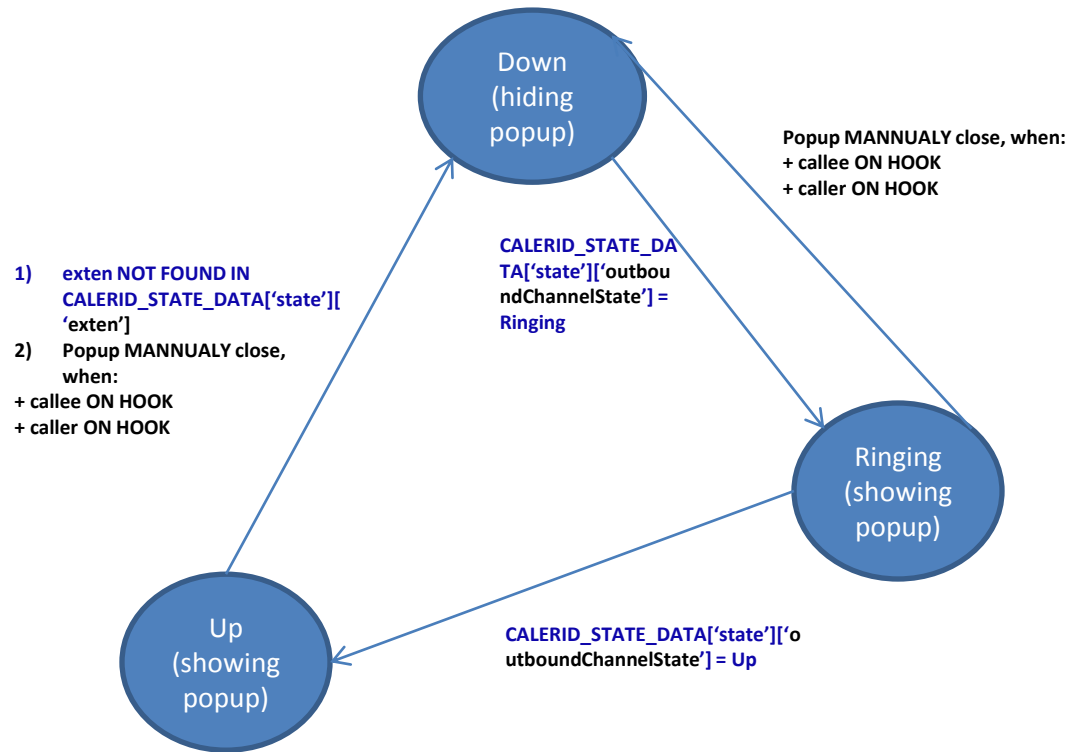
Update notes to DB



Connections between Socket clients, Socket server, ARI client



Socket client states



Socket clients, Socket server authentication

