PBX specification/requirement document:

We need to develop one or more boxes which should contain the following functions and or if there is any other required during the discussion till the final documentation. The whole project will not take more than 5 days including testing and bug fixing. All the bugs should be sent to authorize persons only who will take required action to request a solution to the problem. During the whole implementation procedures should be followed by service providers in order to avoid any time breaks. Communication between provider and buyer is a major and critical issue which should not be very slow, could be based on telephony talk or any other. Below are some basic details of required functionality for IP PBX.

Software/ Hardware

Operating System	: Linux/CentOS or any other
Version	: As needed
Hardware	: Dell Power Edge 2650 2Ghz Xeon Processor 2 GB Memory
Ethernet	: Gigabit Ethernet hardware
Bandwidth	: Can be discussed

Asterisk and OpenSer can be used with or without TLS which should be discussed before the start of this project.
We could request for a secure voice communication system.

IP PBX Detailed Specifications:

It is not an open IP PBX and will be used for personal purposes. These requirements are basically based on a working implementation.

Users:
System administrator should be able to cerate, delete and add extensions (SIP). Admin should be able to create extension based on number of digits e.g. the system will ask while creating the user if it should be 4 or 6 digit extension. Here is some important point that this extension is allowed or not to call other extension numbers in the system.

Basically we will create one account for each user and each user will expect number of extensions in same user group.

- Group users will be allowed to call other member in the same group but not other groups.
- Each user should be able to have the web GUI to set voice mail, call FW and other features.
- *Time based call FW* : Users should be able to set there incoming DID routing on the bases of time e.g. if a user have an incoming number and want to FW calls

only after 5PM then all the calls should be FW to the specified number after 5PM till mentioned time.

- Each user have a DID number which will be allocated to user by admin. Which also means that there are a lot of DID numbers available and routed to the machine.
- All the codecs and codec translation should be available which would include G729.
- Voicemail to each extension will be saved and can be accessed through one DID which can be called from outside the system or inside the PBX. Basically is a voice mail box with password. All these voicemails should also be mailed to the user on specified email ID in the profile.
- User should be able to call any destination in the world, we will provide the VOIP provider interconnection details.
- Each user will be billed based on the call rates which we will provide. All the users are post paid. User has a login where he can see his/her call detail report.
- Call waiting, call hold with music and without music, call FW, call waiting and call conferencing should be available.

Users are able to login using the web and can change there web passwords which are not the same as SIP logins. It means that users will be able to change there passwords for web but they are not able to change there passwords for there SIP logins and only the admin has the options to change the password.

Call routing: local and international format

This is the most simple because all the calls made to the same country of origin will be passed on to service provider and if user is calling on a local number then user will not be using the country code. We have here some work to do only for one country for example if the PBX is used by some group in India then the will not use country code all the calls will be converted to international format and will be passed on to service provider. For example for India CC is 91 and user group is in India then user will not dial 91 before the local mobile number or landline of India system should be able to know or will ask when a group will be created that if this group members will be using local number without CC then the system will have to add CC and pass it on to provider.

Group Name : ABC
Country : India
CC :91

Group user : 1212
User will dial a local number in India for example 0254587487, then the system should know that there is one zero before the number and will replace this one zero with 91 or 0091.

Registration: Under this, we will provide a registration interface, where users will come and fill the registration form. After submitting the registration form, an activation email will be sent to the email id given by the user, once the user confirms his email, his PBX account will be setup and activated and he can login to his account.

IVR

An IVR is required to be set by each user or each group. Each user IVR should be only one and a group can have up to 5 IVRs. A group may be using a single DID to play an IVR to tell the user that you have entered in group ABC and if you have a extension number then please dial or hold for the operator.

GUI

A simple GUI is required on a different IP then the SIP login IP/domain/alias. User should be able to login to there accounts and can download/listen there voice mails set call forwarding (charges will be applied).

Inside group calling (yes/No)

Admin should have the right to allow users of each group to call on there group extension numbers. Means admin will allow if a user can dial an extension number or not, if not user will hear a response that you don't have this facility or any other system response voice.

VoiceMails

Users of each Group can also voice mails on there mentioned mail ID's. They can set it in there web login. Each voice mail should have a limit of seconds or minutes. It is important to keep the best voice quality and while saving the space on server.

DIDs

DID numbers will be routed as needed to the same box to allow each group/extension to have one DID if required.

- The above motioned requirements need to be discussed if one can understand them technically.
- Full documentation is required at the end of this project for future references.