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Secret Phone

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Abstract:
Voice Over Internet Protocol (VOIP) carries telephone calls over LANs and the internet with IP, there is no wasted capacity as this is with circuit Switching. This reduces cost. An android app which can be downloaded in smart phones for making free and secured calls. This Application is mainly based on VOIP, SIP (Session Initiation Protocol) and PTPP (Peer-to-Peer Protocol). An app does not inquire any special charges except internet charges for making calls (if WI-FI, then call is free).
In this project the server is required only for establishing the call. Once the call is connected and user start their communication the server is disconnected automatically. Thus the main advantage of this app is that the data of the calls are not stored in the server, hence the call is very secure. This application is built on android platform using Asterisk server as the backend. The Asterisk Server is an open source and it runs on Linux platform.

1. Introduction

The purpose of this detailed design is to make calls using internet in and across the world. Our system is a peer-to-peer system and thus the data is directly passed between the two users without getting stored in the server so the call made will be secured. For this purpose we have build an application which runs on smart phones. Android being the developing the trend have been used to build this app. The call is not charged from the main balance but required mobile internet, if Wi-Fi is available then the call is free. In addition to the calls messages can also be send. Notifications such as missed calls and the history of the calls is also available in this app.

In the existing system, the data of the call is passed through the server and hence it gets stored in the server. Thus it can be trapped easily. So the ultimate purpose of this new system is to provide secured calls that cannot be trapped by any malicious users.

2. Existing System

Currently the calls which are made through the internet are not safe. The server is essential to make calls and the calls will not be in connection if the server gets disconnected. The data are passed through the server when the call is made and so the data are stored in the server. This storage of data in the server leads to hacking. The apps which are already existing consumes larger memory space when installed on any smart phones.

3. Drawbacks of Existing System

- Data can be trapped, because data is passed through the server.
- It does not run on an open source.
- User has to log in every time to make the call.
- More memory space is required.

4. Proposed System

This mobile app runs on an open source server. The call is established and made Peer-to-Peer once the communication is started between the two users and since the data is not passed through the server the data cannot be trapped. User need not log in every time instead they can just dial the extension to make the call. It requires less memory space when installed. As far as the mobile app is considered the calls can be made worldwide where the call cost is same for both local and international calls.

5. Modules

The three modules involved are:

- Building Asterisk Server
- App creation module
- Amalgamation module

6. Building Asterisk Server

The Building Asterisk Server module will express about two main phases that are being involved which are as follows:

i. Asterisk Installation: The Asterisk installation is the phase where packages such as Dahdi and Libpri are installed in the Linux server machine. These two packages are need to support the communication establishment.

ii. SIP Configuration: The SIP will enable call as it allows the server to maintain details of the callers, their username, password and unique extension given to each user. Thus the user can make calls by dialing the extension given in the server.

7. App Creation Module

The App Creation module explains about the following features which are

- The App Contains a simple dial pad. This dial pad contains the digits from 0-9 and the special characters which are * and # in a grid structure.
- Along with the dial pad, there are two icons which is used for accepting the call and for declining the call.
- It also contains features to add details about the contacts, we can store the contact name, their extension and other details related to the contact.
- The history of the calls can also be viewed. The history contains details about the incoming calls, outgoing calls and the missed calls.

8. Amalgamation Module

The Amalgamation module is the module that is used to connect the App with the Asterisk Server. This module has the following features:

- This module is used to create an account for the user. The SIP should be the type of the account that is to be chosen.
- The details for the account should be given as same as what is given in the server.
- The details that are to be inserted in this module are username, password and the outbound proxy. The outbound proxy is the IP address of the Asterisk server.
- When there is proper network connection in the server and if it is connected to the mobile, the registration of the account will be successful.

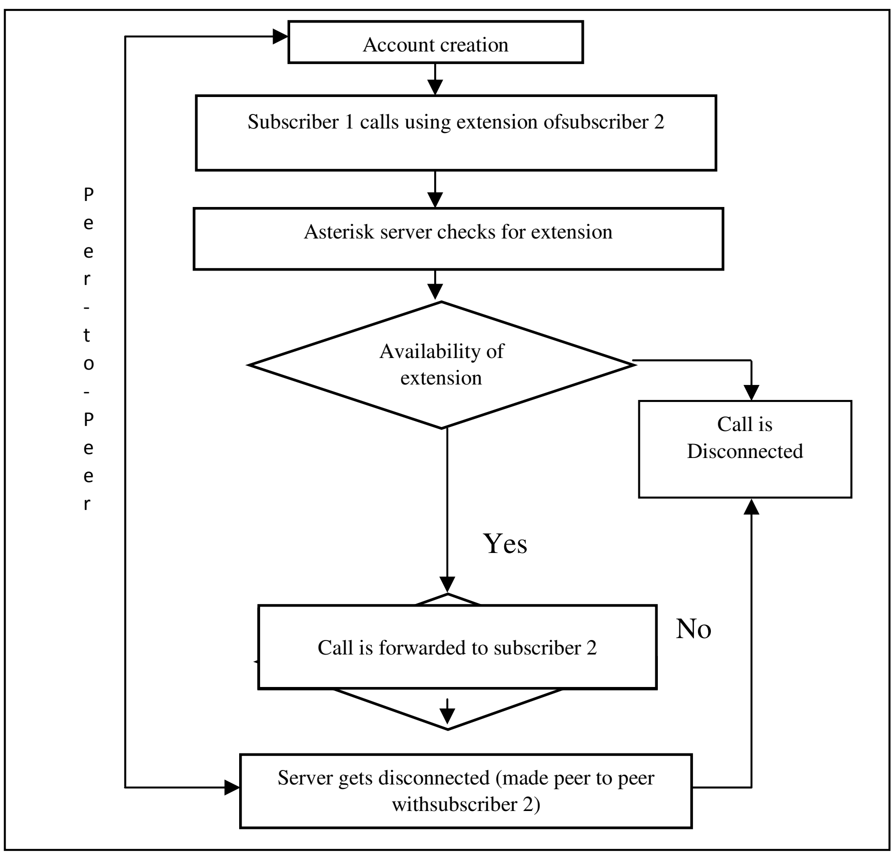

Figure 1: Data Flow Diagrams

Data flow diagram is used to describe how the information is processed, stored and identify how the information flows through the processes. Data flow diagram illustrate how the data is processed by a system in terms of inputs and outputs.
Data flow diagram is made up of number of symbols,

- Squares representing external entities, which are sources or destinations of data.
- Arrows representing the data flows, which can either be electronics data or physical items.
- Open-ended rectangles representing data stores, including electronics sources such as databases or xml files and physical stores.

9. System Specification

9.1. Hardware Requirements

These are the hardware requirements for the system,

- Processor :Dual core 2.70GHz
- RAM :512 MB or more
- Internal Memory :512MB or more
- Device :PC(windows os) & Linux machine

9.2. Software Requirements

These are the software requirements for the system,

- Dahdi
- Libpri
- Eclipse
- ADT(Android Development Tool)
- SDK(Software Development Kit)
- AVD(Android Virtual Device)

10. Database Design

Designing the database is the part of the system design. Data elements and data structures to be stored have been identified at analysis stage. They are structured and put together to design the data storage and retrieval system. A database is a collection of interrelated data stored with minimum redundancy to server many users quickly and efficiently. The general objective of database design is to make data access easy, inexpensive and flexible to the user.

USER NAME	PASSWORD
User 1	abc123
User 2	def123
User 3	ghi123
User 4	jhk123
User 5	mno123

Table 1: SIP Configuration

USER NAME	EXTENSIONS
User 1	6001
User 2	6002
User 3	6003
User 4	6004
User 5	6005

Table 2: Extensions

11. Input Design

The input design is the part of overall system design which requires very careful attention. If the data going into the system is incorrect then the processing and the output will magnify the errors. Input design features can ensure the reliability of the system and produce result from accurate data or they can result in the production of erroneous information. In this application, the input given to the system is the extension of the caller as given in the server. This extension is used to compare and check the extension in the server and thus the call gets connected between the two users.

12. Output Design

The output design presents the manipulated data to the end user. The output design acts as medium of communication to the user by providing the desired data. A quality output is one which meets the requirements of the end user and presents the information clearly. The output of this application is the connection of call from the caller to the person he calls. The successful connection of call even after disconnection of the call is the output.

13. Screen Shots

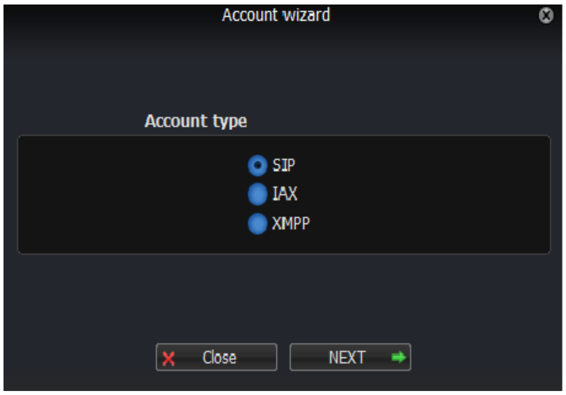

Figure 2: Protocol selection : Due to small bandwidth Session Initiation Protocol is being selected

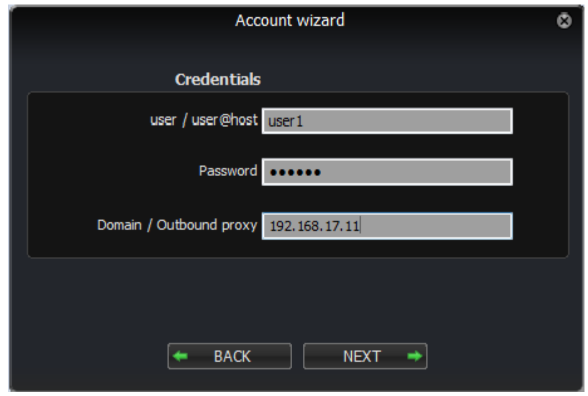

Figure 3: Setting up password: For security purpose password is being setup and for easy identification IP address is given

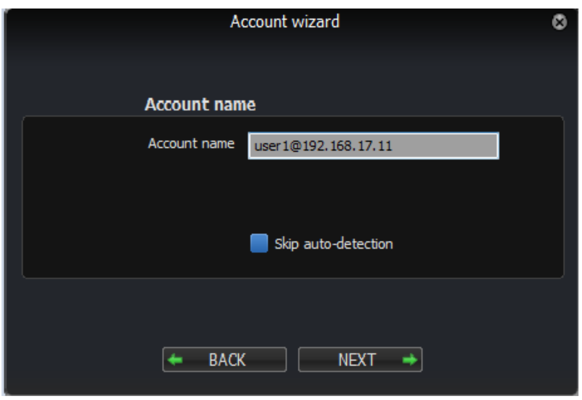

Figure 4: Account Creation: Create an account name with IP address

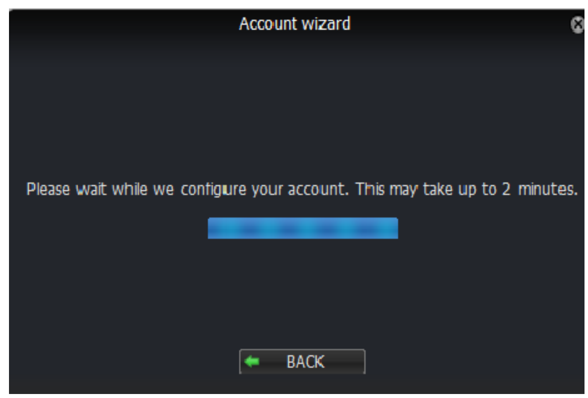

Figure 5: Verification of account: The IP address, password and user name is verified

Figure 6: Call dialing: To make a call dial a number in a dial pad

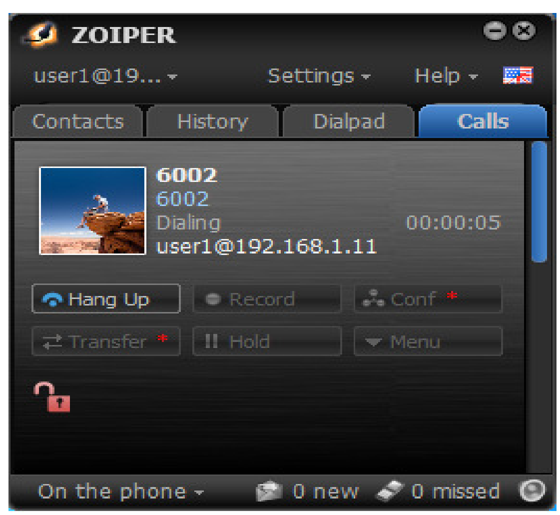

Figure 7: Server-to-client connection: Once call is connected server-to-client connection has been disconnecte

14. Conclusion

This type of project can be used to make free calls which is of low cost and that is secured. The main advantage of this project is that once when the call gets connected, the system is made peer-to-peer and thus the server is used only for call establishment so that none of the data is stored in the server. Hence the call connection is secured.

15. Future Enhancement

The call made can be only audio calls. Other advanced features like making video calls and the features to send text messages through the App have to be developed. So far the developed app can be used only inside an organization through intranet due to the complexity in adding the app in play store as itinquire special charge to launch ourapp.Thus this app has to be modified to make it work by using internet.

16. References

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