WIPRO
Applying Thought

Integration of GSM Module with PC Mother Board (GSM Trunking)

WHITE/Technical PAPER

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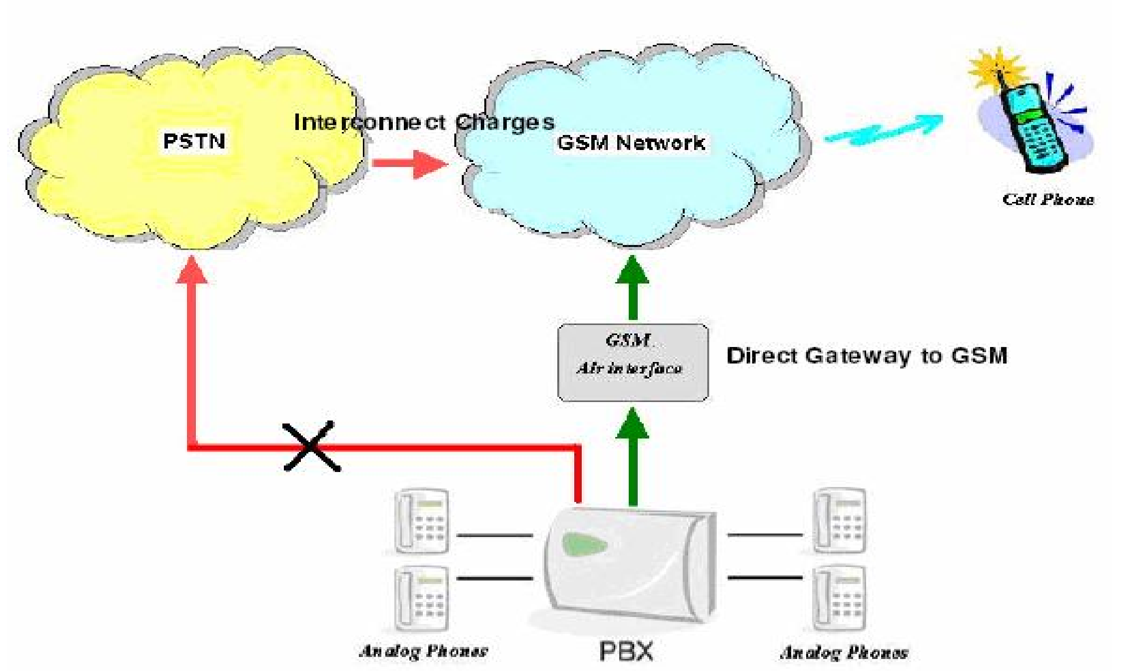
1. Abstract

The basic idea behind this paper is to integrate GSM module with PC mother board. The GSM Gateway (trunk) provides GSM Air interface facility to the IP PBX (Asterisk). Making outgoing calls to any GSM mobile/Landline from any registered extension of IP PBX (Asterisk) will by-pass costly PSTN line.

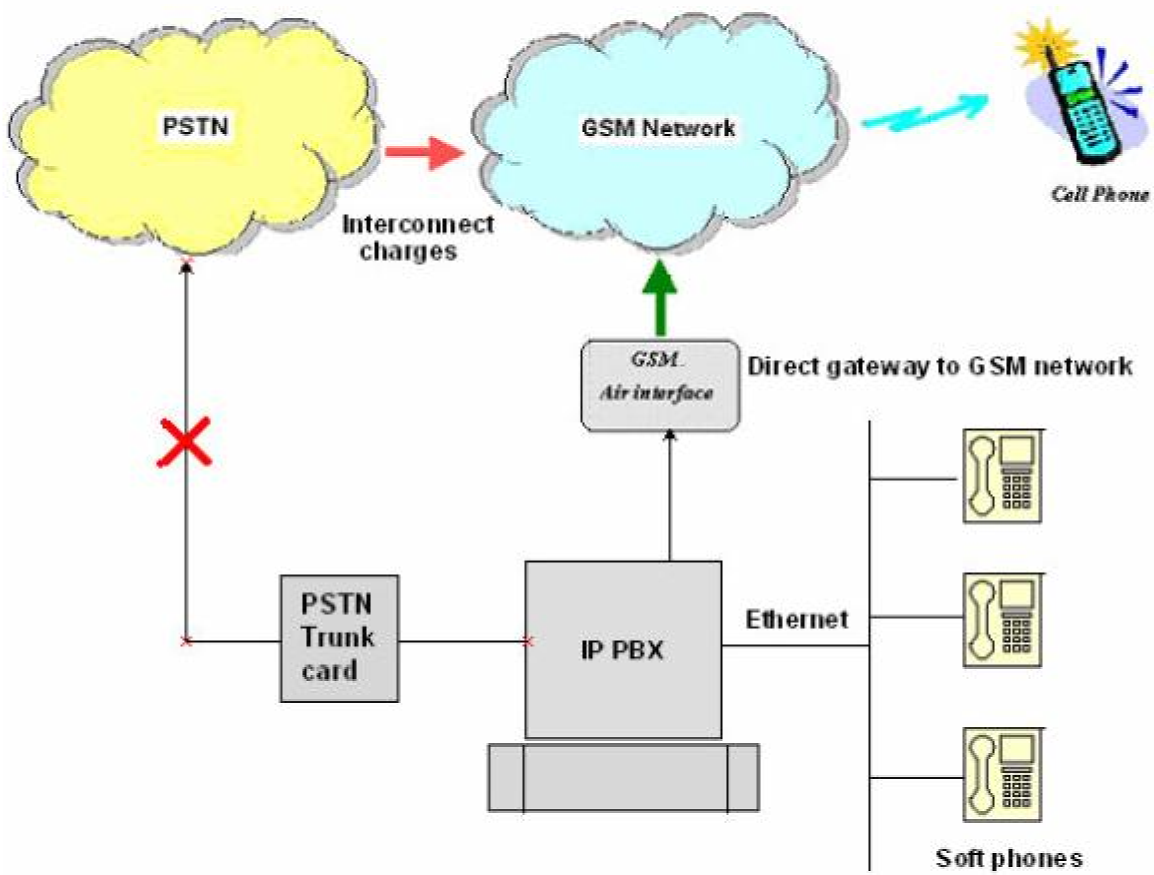
2. Introduction

Today, the mobile market has overtaken the fixed landline installations. While most organizations have traditional fixed landline phones, individuals own mobile phones. When a landline user from an organization calls to a mobile user, the call is routed from the organization's Private Branch Exchange (PBX) to the PSTN network, which in turn routes it to the GSM network. This accounts for the additional "interconnect charges" between the PSTN and GSM network, which are passed on to the organization. Thus the calls from landline to cellular networks are costlier than the calls within a GSM network. The project aims at making landline to mobile calls cheaper by exploiting the difference in the costs of landline-to-mobile and mobile-to-mobile calls. The basic idea behind the project is to provide a GSM air interface to the PBX that would serve as a gateway (trunk) between a GSM network and a company's PBX. All the outgoing calls to mobile phones can be directed to the GSM interface by the PBX. Thus a landline-to-mobile call will effectively be a mobile-to-mobile call. As depicted in below Figure, calls to mobile number, originating from the organization will go through a GSM air interface at the PBX thus bypassing the PSTN.



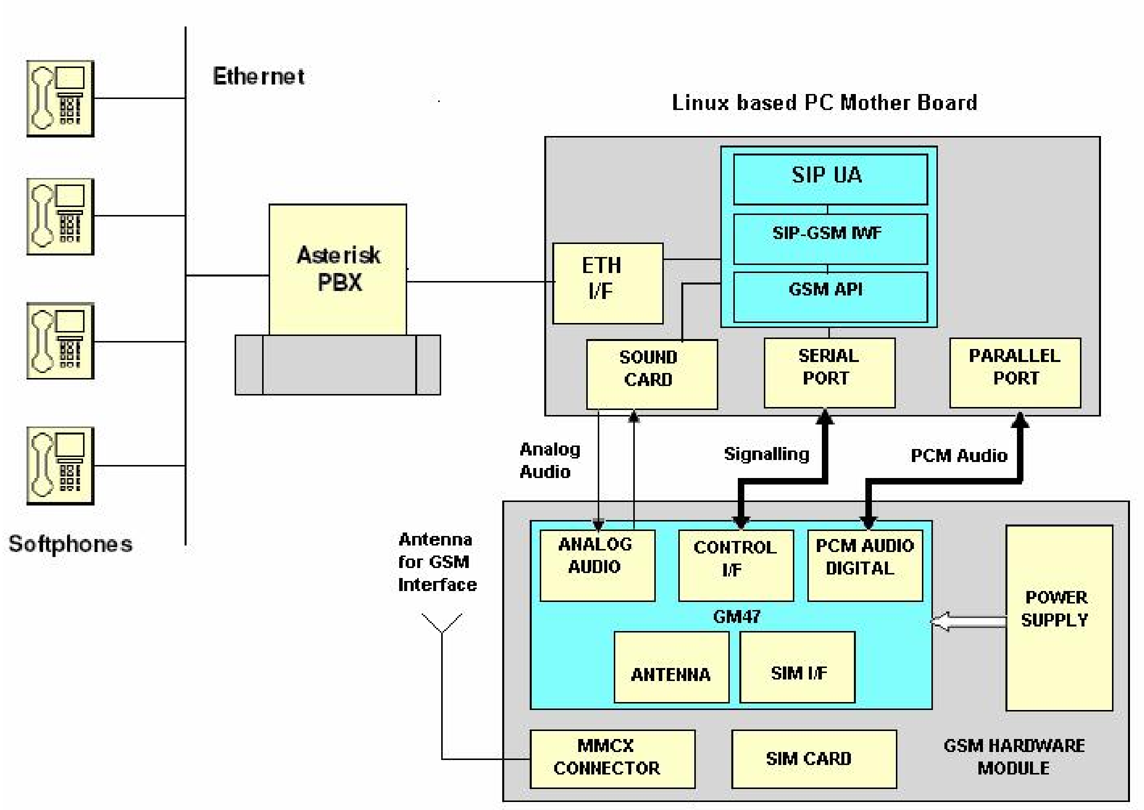
3. Proposed System

Many organizations have deployed a Voice over IP (VoIP) network with an IP PBX for call routing. Such networks access the fixed landline phones or the mobile phones through a PSTN trunk or a dial-up connection at the PBX. So the calls from the IP network to mobile numbers still goes through a PSTN network resulting in interconnect charges. We have applied the concept of GSM trunk to a VoIP network. Our aim is to make VoIP users get connected to GSM network directly, bypassing the PSTN interface between the VoIP and GSM network. As shown in the figure below, when an IP user calls to a mobile number, the call will be directed by IP PBX to the GSM network through a GSM air interface instead of the PSTN trunk. A similar concept can be applied when a mobile user wants to call a person in the IP network. The call can directly land to the GSM trunk and then routed in the IP network.



4. Solution Description

The system consists of an IP network with several soft phones connected to the soft PBX, Asterisk. The below figure shows the PC bother board and GSM hardware board. The GSM hardware board has GSM module, power supply, SIM card interface and serial port to configure GSM module from the host PC.



SYSTEM BLOCK DIAGRAM

Whenever a user in the VoIP network dials a mobile number, the Asterisk forwards the call to the SIP User Agent (UA) running on Linux based PC, which supports SIP-to-GSM inter-working. It converts the SIP call to GSM call through the GSM hardware interfaced to PC mother board. Once the call is established, the voice data is routed between the caller and the destined mobile. Similarly when a call is made to the GSM module from an outside mobile number, it will be routed to a fixed extension in the IP network.

4.1. System Specifications

- Development of GSM interface board with GSM module and serial interface to communicate with Linux based PC.
- Porting of SIP User Agent (UA) on Linux based PC.
- Development of SIP-to-GSM Interworking function (IWF) for GSM call signaling
- Modification of SIP UA control plane to support SIP-to-GSM IWF.
- Setting of Asterisk Dial Plan for call routing to GSM trunk.

5. Implementation

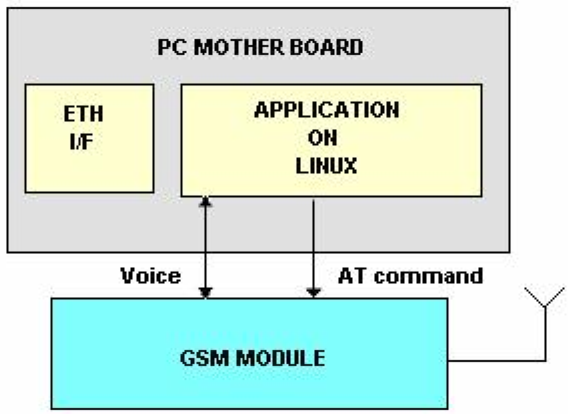
5.1. Product Requirements

PC mother board as working platform:

The PC mother board has been used as development platform for this project.

Development of GSM interface board:

We have to provide additional functionality both in terms of hardware and software to make a GSM trunk around Mother Board. The hardware will comprise of a GSM module that can be controlled through standard modem AT commands. The figure below illustrates the concept where the GSM interface is added to Mother Board through a GSM interface board. (The interface includes both voice and control path).



PC MOTHER BOARD - GSM BOARD INTERFACE

SIP Protocol:

The Session Initiation Protocol (SIP) is a signaling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. Over the last couple of years, the Voice over IP community has adopted SIP as its protocol of choice for signaling. SIP is an RFC standard (RFC 3261) from the Internet Engineering Task Force (IETF), the body responsible for administering and developing the mechanisms that comprise the Internet. SIP is still evolving and being extended as technology matures and SIP products are socialized in the marketplace.

5.2. Hardware Components

Linux Based PC:

Linux based PC used to interface with GSM hardware board. PC serial is used to control the GSM module. And also Linux based application which supports SIP-to-GSM inter-working.

GSM hardware board:

The GSM hardware board has been developed to and would be interfaced with Linux based PC mother board. The board has GSM module, serial port interface to communicate with PC and Power Supply for providing necessary power to the modules (3.6V and 5V).

5.3. Software Components

Asterisk soft PBX:

Asterisk is complete PBX software compiled on Linux. The IP network based on SIP protocol is built by defining extensions in Asterisk. It routes calls within an IP network based on a dial plan. It also acts as a SIP proxy and SIP Registrar in the IP network.

Asterisk allows you to craft a telephony system to address specific requirements. It does this by providing a library of basic telephony functions which you then use as script building-blocks. Calls into the system trigger these functions through digit patterns (referred to as extensions), gives complete control of complex call routing concepts with relative ease. Common PBX functionality such as voicemail, call queuing, conferencing, music on hold and others are all included. Asterisk is one of the few PBXs in existence that connects legacy telephony technologies such as PRI or Analog trunks through the same switching logic as state of the art VoIP interfaces such as H.323 or SIP.

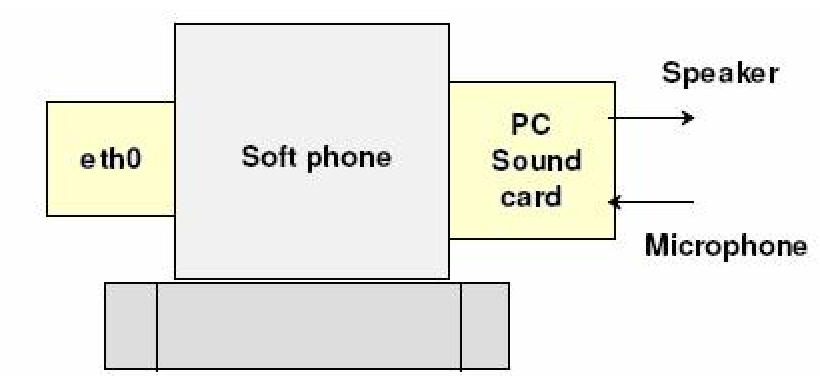
SIP User Agents:

These are the endpoints in the SIP call and can be hardware or software IP phones. These user agents comprise of two planes

a - Call control/signaling plane – This handles the call establishment and termination. This is done using SIP protocol.

b - Media plane – This handles the audio data to and from the sound device.

The SIP User Agent can register its presence to the SIP Registrar by sending registration packets. SIP UA by default make use of PC sound card for voice transfer as shown in below Figure.The analog audio input from MIC is digitized by PC sound card, packetized by SIP UA and sent to the destined caller over Ethernet. Similarly voice data coming from the destined caller in Ethernet packets is de-packetized and sent to PC sound card. The sound card converts it to analog form and sent to the speaker.



SIP SOFT PHONE ON PC

To make a VoIP network, few soft phones are installed on desktop machines and they register to Asterisk PBX. SIP UA ported onto Linux PC, is a web phone called "**Linphone** " The Linux based softphone version used is **Linphone-1.1.0.** The choice of this SIP UA from several available ones as Linphone is available in console version (called linphonec), which is helpful for our embedded needs. Linphone has an easy to understand C code and offers itself easily for possible modifications.

AT Commands:

These are control commands that allow call setup; call control etc. in a modem from a controller (e.g. PC Hyper Terminal) through serial port. For each command the modem returns responses like OK, ERROR, NO DIALTONE, etc. to indicate whether it was successful in executing the command and its status after the command.

Some examples of AT commands are:

Start a voice call **ATD9886098860;**

Start a data call **ATD9886098860;**

5.4. GSM Hardware Module:

The GSM hardware module integrates the necessary hardware needed around the Linux based PC mother board to make it function as a GSM trunk.

The daughter card consists of the following:

- A GSM module with necessary interface circuit such as SIM card interface.
- Serial interface.
- Audio interface (MIC and Headphone Interface).
- Power Supply Circuit

5.5. Software Implementation

GSM API:

Some basic GSM APIs were developed to control the GSM modem from the Linphone application. These APIs involved sending a command to the serial port. After each command is sent, the response from the modem is read back. This response tells whether the modem executed the command successfully and the status of the modem after the command.

SIP-to-GSM Inter-working Function (IWF):

The aim of this IWF is to control the GSM modem based on requests from the SIP end and to reply back to the SIP end based on responses from the GSM modem. When Linphone receives a SIP INVITE to a mobile number, the IWF will start a call to the mobile number by sending an ATD command (if the modem is in READY state). It then polls the modem for a RINGING status. When it detects a RINGING status at the GSM end, it issues a SIP 180 Ringing to the caller. The IWF then polls the modem for CONNECTED state. When the called party accepts the call, the modem goes to CONNECTED state. The IWF then issues a SIP 200 OK to the caller that then starts the media flow between the two parties. When the caller wishes to terminate the call, it issues a SIP BYE. On receiving a SIP BYE, the IWF issues a Hangup command to terminate the call.

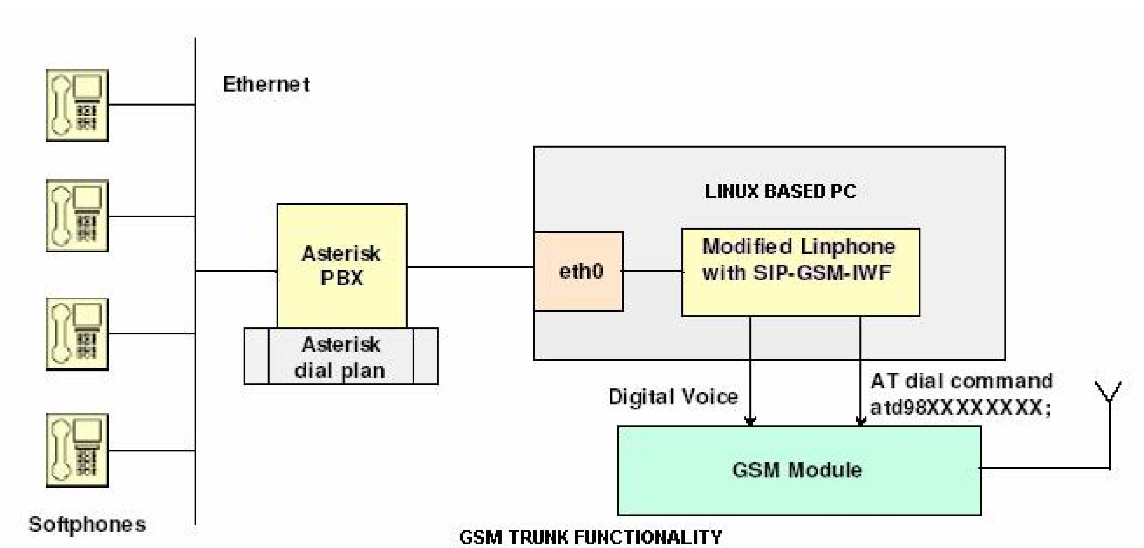
Adding this IWF support in the existing Linphone involved identifying the potential areas in the source code from where AT commands can be issued whenever a SIP message is received.

Asterisk Dial Plan:

The Asterisk IP PBX resolves the dialed extension number to a SIP URI by means of the dial plan. The dial plan tells Asterisk which UA in the IP network must be called when a particular number is dialed and then sends an INVITE message to that SIP UA on behalf of the caller.

Operation

The call flow achieved is as shown in Figure below. The SIP user in the IP network calls to a mobile number. Based on the set dial plan, Asterisk routes the call to the Linphone SIP UA running on Linux based PC. It also inserts the destined mobile number in the SIP INVITE header. The SIP UA supporting SIP-GSM IWF extracts the mobile number from the header and then starts a GSM call. Similarly, when an incoming call is detected on the modem, a fixed extension in the IP network is called (the operator of the IP network).



6. Conclusion

The concept of GSM trunk proven in the project can be easily deployed in any existing VoIP network. This concept further encourages the VoIP networks with a GSM trunk and a PSTN trunk (T1/E1 line). Thus, a network can be formed with access to any of the three networks VoIP, GSM or PSTN and take advantage in terms of call charges and other benefits provided in each. Also multiple GSM trunks can be provided, each supporting a service provider (Airtel, BSNL, Hutch) and make calls still cheaper. Other features that can be added are least cost routing, call forwarding from one network to the other.

7. Acknowledgements

I would like to take this opportunity to thank my colleagues (Arshad, Mohammed, Rama krishna Raju, Nataraju.K, Srikanth) who have helped in at various stages of this paper.

8. References

GSM & SIP:

GM47 Integrators Manual & data sheet.

RFC 3261 (SIP)

Links:

http://www.sonyericsson.com/
http://www.asterisk.org/
http://www.voip-info.org/wiki-Asterisk

9. Glossary

Terms	Definition
URI	Uniform Resource Identifier
API	Application Programming Interface
AT	Attention (Hayes Command Set)
GSM	Global System for Mobile Communications
HTTP	Hypertext Transfer Protocol
IETF	Internet Engineering Task Force
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
RFC	Request For Comments
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SIM	Subscriber Identity Module
UA	User Agent
VOIP	Voice Over Internet Protocol

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