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ACCESSING PSTN INTELLIGENT NETWORK FUNCTIONS FOR A DATA CALL

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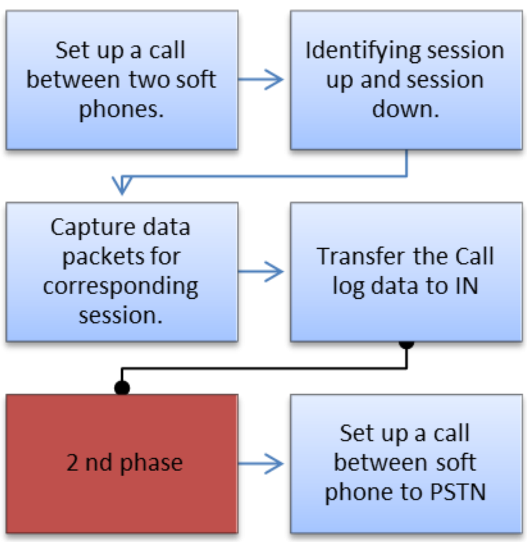
Graphical abstract



Abstract

As Telecommunication has become a basic requirement of the today's society there had been a vast increase of competition between the telecommunication service providers. Every service provider tries to cater the customers the best services for the lowest charges. Customers have the expectation of receiving the most recent technologies for the cheapest cost. Triple play (voice, data, content) is becoming everyone's requirement today in communication. Voice over IP (VoIP) comes handy in considering Triple play. The design presents in the paper is an approach developed for a company that holds an Internet Service Provider (ISP) license from the Telecommunications Regulatory Commission of Sri Lanka (TRCSL) for island-wide coverage. The said company is a subsidiary of the leading Telecommunication provider of the nation. The problem faced here by the subsidiary is finding the capital cost to provide this highly exclusive IN (Intelligent Network) services to the customers in their initiation phase to cope with the existing competition. As an optimum solution, came up with the idea of accessing existing legacy PSTN core network of the leading telecommunication service provider which enhances most of the highly exclusive services. The system presented in the paper is a small model of an existing network that allows online billing facility for data calls as an IN facility.

Keywords: VoIP, PSTN, IN, internet service provider, SS7, PBX, IP

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1.0 INTRODUCTION

In the current context data communication plays a vital role in everyone's life and in every day the industry is becoming immensely competitive requiring value added services from the service providers for the minimum most competitive cost. Thereby today challenge for the service provider's is to be distinctive in the market by providing the new innovative attractive solutions by the usage of existing capabilities available with them.

The model discuss here is an approach in using accessing PSTN (Public Switched Telephone Network) network IN (Intelligent Network) functions available with the mother company, Sri Lanka leading telecommunication service provider for the state for a subsidiary to facilitate and provide value added services. The model is developed to provide online billing service for the subsidiary for their data calling facility. The subsidiary holds an Internet Service Provider (ISP) license from the Telecommunications Regulatory Commission of Sri Lanka (TRCSL) for island-wide coverage same time holds a spectrum for

WiMAX (Worldwide Interoperability for Microwave Access) operations.

The traditional PSTN of this telecommunication service provider is a circuit-switched network in which analog voice data is transmitted across the transmission medium [3]. The medium used in the PSTN is copper and fiber-optics. It also comprises of a hi-tech IN. The IN as a whole allows service providers to differentiate themselves by providing value-added services in addition to the standard telecommunication services in the legacy PSTN [3].

IN [4] is worked on the platform of Signaling System 7 (SS7) protocol. In SS7 signaling control information (dialled number by the caller, other relating details of the caller) is exchanged in packet switched (use dynamic routes for transmission) format [5]. Some of the highly developed services provided by the IN are Telephone number portability, Televoting, Call screening, Toll free calls, Prepaid calling (online billing) , Account card calling ,Centrex service (Virtual PBX), Private-number plans (with numbers remaining unpublished in directories) and so much more [5]. The problem faced here by the subsidiary is finding the capital cost to provide this highly exclusive IN services to the customers in their first phase. In the path of finding the optimum solution the idea came up as using the existing PSTN core network of the leading telecommunication service provider which enhances most of the highly exclusive services.

The company's basic requirement at the launching stage is to facilitate the customers at least with online billing facility. To achieve the above mentioned task it is required to convert the IP based data to another form which can be identified by the IN of the PSTN structure. For online billing in standard PSTN structure computerised record is generated at every exchange for all the calls that are parsing through it which is known as CDR (Call Detail Record) [3], [6]. Thereby in our approach we try to convert IP based data to a compatible form which can be identified by the PSTN of the leading telecommunication service provider and also to produce CDR format files to grant online billing facility.

2.0 RELATED WORK

The interconnectivity of IP network to PSTN (SS7 signaling mode) to give out VoIP[1] services to customers is a hot topic discussed in telecommunication sector. VoIP has been implemented in various ways using both proprietary and open protocols and standards. The most extended are H.323 [7] and SIP [2], [8]. Telecommunication providers have over the years have gone ahead with the discussed topic in connecting VOIP to PSTN network by developing FXS(Foreign Exchange Station) and FXO(Foreign Exchange Office) gateways [9],[10]. Basically there can be three ways in approaching the above mentioned task. These three methods have their own issues and challenges but according to our requirements and objective can adapt one of the methods.

I **Distributed Infrastructure Model:** In this architecture ownership infrastructure is divided by outsourcing some of the components to the outside of service provider. Basically the outsourced are control of signaling and network elements making it a distributive structure. This distributive service model is advantages to reduce capital expenditure costs and the risk of investment in new and changing technologies and vendors. According to VeriSignSIP-7 Service provided in North America for interconnectivity of SS7 and SIP based network for the objective of accessing intelligent services is a classic example for distributive infrastructure [11]. In this approach single SIP feed is used to meet the requirements. In this service internet service provides the media gateway while all the necessities expertise to administer, maintain, and support the more complex and costly elements the soft switch and SS7 signaling. are granted by SIP-7 Service provider.

II **Fully Outsource Model:** In this model third party manages and provides all the elements. Advantages where a quick startup is required also significantly reduced need of experienced staff in IP and SS7 technologies. This model can be disadvantages as the service provider get locked with single-vendor solution.

III **Fully Insource model**: In this structure the service provider does all functions as purchasing, implementing, and managing of all elements (Soft switch, Media Gateways, and SS7 Signaling) required accessing the PSTN. Thereby this though the above two methods are less hazel in operation for the subsidiary the fully in source method was the only effective model when considering the availability of resources with the mother company and cost at the initiating phase.

3.0 METHODOLOGY

Final target of this paper as mentioned above is to access legacy PSTN network facilities to cater for IP calls for the subsidiary. By achieving that target the objective is to access IN of PSTN of the mother company and grant online billing as a value-added service for data calls taken by the subsidiary users. For the feasibility of approaching the final target the flow is divided to four major steps as in Figure 1.

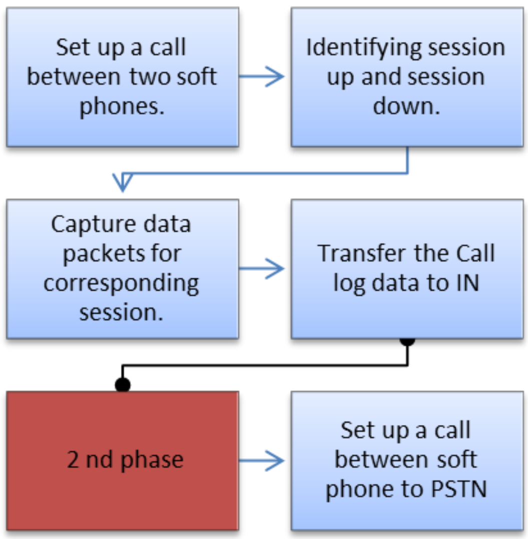

Figure 1 Process flow of the steps

1. Set up a call between two softphones:

In this phase main idea was to implement a VoIP network as a test model. In the process of implementing the VoIP network a PBX (Private Branch Exchange) [12] and softphones are required. The main function of a PBX is to work as an exchange for telephone systems in small scale (intercom as in enterprises). PBX consists mainly of several branches of telephone systems and it switches connections to and from them, thereby linking phone lines. Companies use a PBX for connecting all their internal phones to an external line. This way, they can lease only one line and have many people using it, with each one having a phone at the desk with different number. The number is not in the same format as a phone number though, as it depends on the internal numbering. Inside a PBX, you only need to dial three digit or four-digit numbers to make a call to another phone in the network. We often refer to this number as an extension.

A softphone is a software program which functions equal to the traditional phone. Softphones are used on making telephone calls over the Internet (skype, google talk are some examples). Establishing a call between two softphones are similar in the function to the process of call set up between two traditional phones where as in a call between two softphones happens via an IP network.

Three basic requirements was followed for the establishment of the VoIP network.

a) Installing softphone clients on a Windows based PC.In the test model Xlite is installed on Windows Operating system in two different computers via a Ethernet network
b) In another PC to be installed with an operating system compatible to PBX server running. For the purpose Red Hat Linux 04 was installed on a PC.
c) PBX is installed on it and configured to meet up the requirements. Asterisk 1.6.2.7 is installed and configured to work as the PBX. Asterisk [13] is an open source VoIP server which can offer communication channels using almost all the VoIP protocols including SIP.

2. Identifying session up and session down:

In the attempt of identifying session up and session down had to deal with SIP (Session Initiation Protocol). SIP has been standardised by the Internet Engineering Task Force (IETF) for creating of VoIP connections. SIP is a signaling protocol for creating, modifying and destroying dialogs between multiple endpoints [14].
Instant Messaging sessions, Phone calls over the Internet, Gaming servers, Resource Location are some of the sessions enable SIP signaling on controlling. specification, each with different purposes. SIP is an application layer protocol based on the client/server model. In the protocol two nodes communicate throw a series of textual representation request-response messages. SIP represents the signaling part of the call and it doesn't carry the voice packets which are transmitted over UDP(User Datagram Protocol) [15]. SIP signaling commands are sent in clear text between the clients and the server to establish the media channel link and only after that point the media will be transmitted from the caller and the called clients as in Figure 2. In the test model JAIN SIP is used which is the standardised Java interface to the Session Initiation Protocol for desktop and server applications .
The purpose is indicated in the field (called method) in each SIP request. The list shows the six methods.

a) INVITE
b) ACK
c) OPTIONS
d) BYE
e) CANCEL
f) REGISTER

Basically, the required types of SIP requests to identify the session up and session down in our approach are "INVITE" &"BYE respectively.
INVITE requests provoke users to participate in a session. The description of the session is enclosed in the body of INVITE requests. This request is used as the session up identifier.

To abandon sessions, BYE requests are used. In two-party sessions, it is implied that the session is terminated if one of the parties abandon the session.
This request is used as the session down identifier.

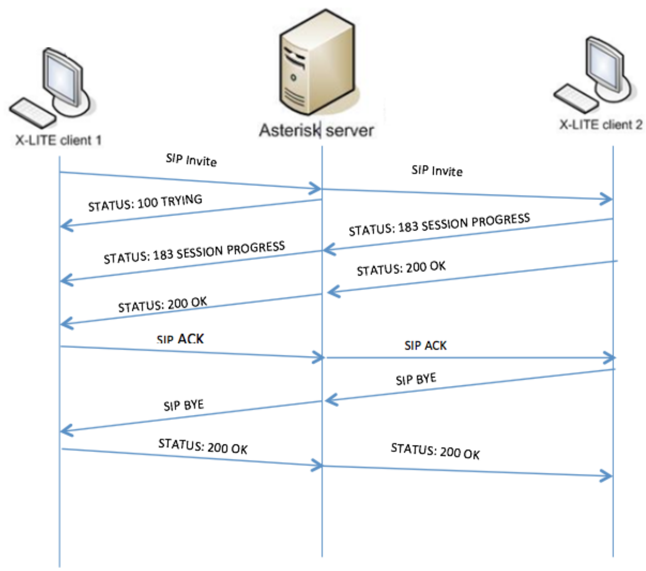

Figure 2 Message flow of two SIP clients

3. Capture data packets for corresponding session:
In the path of obtaining on line billing facility a logged data record of the VoIP call similar to CDR (Call Detailed Record) format must be created. In a telephone exchange a computer record CDR is produced for all calls passing through it and is referred in billing functions. The CDR format require by the system needs to log the caller number, called number, timestamp, call duration. CDR is then feed to the Intelligent Network to make the required an Intelligent functions. INAP (Intelligent Network Application Part) is the application layer which is the application layer at Intelligent network. To achieve the objective data packets are captured and analyzed by the use of pcap (packet capture). Pcap (packet capture) consists of an application programming interface (API) for capturing and analyzing data packets. For the test model Jpcap which is a Java library for capturing and sending network packets is used. With the support of jpcap few steps were followed afterward to log the records of SIP data packets as in Figure 3. By the use of pcap can identify whether the protocol type of the packet is UDP or TCP. Then UDP packets on port 5060 are filtered as a way of identifying SIP packets. By parsing the SIP message string SIP massage is identified. For parsing some characteristics of SIP message is used.
Further the classes and functionalities of the program refer in Table I.

a) Minimum of 9 bytes long data.
b) The first line of data would contain substrings bordered by a single space character. Data lines were terminated by CR/LF characters (/n).
c) A SIP command word such as REGISTER would be the first substring.
d) SIP/ would be the starting characters of the third substring.
Example: REGISTER sip: 10.40.18.177 SIP/2.0/r/n.

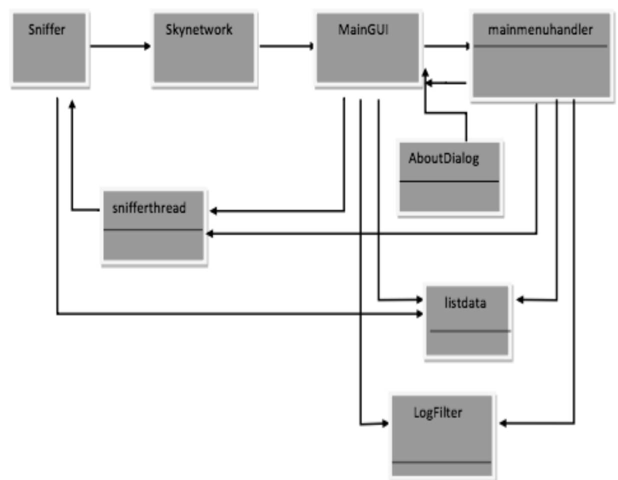

Figure 3 Class diagram of the packet capture

Asterix [16], [17] CDR functionality can be directly supported to transfer the data recorded to CDR format but in the proposed method to achieve the objective data packets are captured and analyzed by the use of packet capturing program modified to cater the requirement (example: pcap/packet capture).

Table 1 Description and functionality of the classes

Class Name	Class functionality
Sniffer	This is the main class of the project which includes receivePacket() method and main GUI(Graphical User Interface).
Skynetwork	Skynetwork connects MainGUI class.
MainGUI	MainGUI includes connection interface which allows selecting network interface and starting the packet capturing
Mainmenuhandler	This handles all the functions of connection interface menu bar.
Snifferthread	This uses thread to synchronize packet capturing and reading processes.
AboutDialog	This is used to display a html file about the project
Listdata	This class is used to list all details of the packet which is captured by the method receivePacket().
LogFilter	This is used to save the listed packet data as a text file.

4. Transfer the call log DATA to TN:
The first three steps of the model is developed by testing data calls to storing and monitoring the logged data (CDR format). The created CDR is then connected to the PSTN Intelligent network to give out the value added functions. CDR for billing is created when a user calls a destination within the IN switch, the

service switch point (SSP) will send a request to the service control point(SCP). The SCP usually responds with the charging and routing needed for the call. When the call ends, SSP will signal SCP that the call has been ended; the SCP will start a sequence of processes which normally include the creation of a call detail record (CDR). This is done by Service Data Function(SDF),the SDF may be a separate platform, or is sometimes co-located with the SCP. In the current IN many platforms are supportive as VoIP with SIP enabling interfaces along with the traditional INAP(Intelligent Network Application Part) for PSTN [22]. Thereby in the approach as we used JAIN SIP to identify the SIP (discussed in step in Methodology) in the same manner JAIN SLEE(Jave based Service Logic Execution Environment) can be further used to [22] coomunicate with the IN as SIP based SCP. Thereby as we have created a record in the format of CDR we can use the same format to feed into IN through JSLEE of the PSTN by recording them in data bases.

With the completion of the discussed four methods the objective of granting the online billing facility can be provided to the customers of the ISP. Further discuss three ways to give the functionality of connecting data calls to PSTN calls as the extended phase 2. Mapping of Session Initiation Protocol (SIP) and the ISDN User Part (ISUP) of the PSTN is the method to achieve the target.ISUP is a level 4 protocol used in SS7 networks. [20] . The function of mapping is done through MGC(Media Gateway Controller)

a) Router for call processing for IP phones, Softphones and PSTN phones: In the current market there are routers which facilitates the function of call processing for IP phones, Softphones and PSTN phones (Example: Cisco CME). The said routers provides the facility to deliver telephony features that are commonly used by business users to meet the requirements of the small or medium sized office. The routers supports on H.323 integration options To control IP phones, analog phones, and faxes. Some routers uses H.323 and the protocol for real-time calls and conferencing over IP(Example :Cisco CME). As per the ITU Telecommunication Standardisation Sector H.323 is a highly Industry Standard specification for transmitting audio, video, and data across an IP network, including the Internet. H.323 is an extension of the standard H.320.

b) ATA(Analog Telephone Adapter) [18]: For a connection establishment between analog (standard and non standard) telephones to a digital telephone system (such as Voice over IP) the device ATA can be used which is also known as VoIP Adapter.

As in Figure 4, ATA can connect only one or two analog phones in a domestic usage non effective and cannot be used for large customer base service providers. For the construction of our small test model ATA was used as the gate way for IP to PSTN conversion.

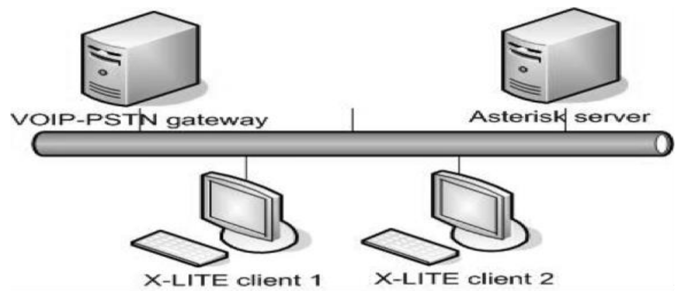

Figure 4 Test model for VoIP to PSTN

c) SIP Trunk [19]: For the traditional PSTN network, the internal IP network connection is provided by the ISP (Internet Telephony Service Provider). In contrast to the traditional telephony (where the Figure 4. Test Model for VOIP to PSTN service provider deliver bundles of physical wires to an enterprise), a SIP trunk allows replacement of traditional fixed PSTN lines with PSTN connectivity via a SIP trunk service provider on the Internet. This offers significant cost-savings for enterprises by eliminating the necessity of local PSTN gateways, costly ISDN(Integrated Services Digital Network) [20], BRIs (Basic Rate Interfaces) or PRIs (Primary Rate Interfaces).If the ISP is in it's initial phase can use this SIP trunks layering which is the latest trend and as mentioned no gateways needed and use a IP PBX in connecting. But for our mentioned subsidiary main target is spending the lowest capital cost and providing the most trendy value added service.

4.0 CONCLUSION

The discussed model was developed only as a prototype for a new service provider in the country to service the customer with the value added services. The prime target of the paper was a creation of a small model that allows online billing facility for data calls as an Intelligent Network facility for an emerging ISP. Also the model was implemented for an operation of 2 soft-phones and through a PBX and a gateway to design the model as the ISP service. By the implementation of discussed model the capital cost for the company will only be for the translation gateways from VoIP to PSTN at the second phase. It was achieved by following four steps as discussed in the methodology. It cannot be guaranteed that the log file CDR will give the most accurate data read when applying the model for a larger customer base due to number of packets parsing through can be very large. As considering it as a matter we made tread methods in the data analyzer for the idea of reducing data packets getting queued. If packet queuing occurs it will be highly disadvantages for ISP in a facility as online billing. In the model the created CDR can be logged in data base and given the access or feed directly to the IN of the mother company (PSTN). With the fed CRD data base of the ISP will be used to facilitate the other available functions of the IN in future to make it a more attractive ISP.

5.0 FUTURE WORK

In the designed model the attempt is to cater customer with the online billing facility but in future the functionalities available with the sophisticated IN functions can be catered as value added services. Further can provide the facilities as in to monitor the last call bill value, Current bill value as an SMS (short message service) etc. Also in the other way around can use the VoIP network to route PSTN users which will result in increasing flexibility, enhanced customer service and dependable call management can be increased. [21] The current trend is abandoning of legacy PSTN and moving to IP based digitized network but in our approach the model is vice versa to cater the requirement but when designing functionalities in VoIP network there are inherited problems in the system which needs to give special attention on QoS(Quality of service), latency, jitter etc [21]. Further at the conversion of VoIP traffic it is extremely require keeping focus on the time-sensitivity of VoIP. Therefore, voice packets cannot be queued and must have priority over data information. If the network is not configured correctly, it will result into sound jittery.

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