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Soft Phone Support Voice and Video Calling Using Sip And Rtp Protocol

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Abstract: *Soft Phone is a VoIP soft phone that uses the Session Initiation Protocol. It is a powerful and unique SIP software telephone that lets users make phone and video calls using single software application using any Voice over IP (VoIP) telephony provider. As a fully featured SIP client, Soft Phone is designed from the ground up to work with current and future IP-based. Soft phone can be used in the place where we need establish the voice and video phone calls in the IP network. This can be installed at any personal computer. It uses the SIP protocol to establish the IP session and using RTP, RTCP and RTSP for media transmission. Using Soft phone they make call to any Soft Phone and Physical hardware SIP phone which can understand the SIP protocol.In computing, a soft phone is a software program for making telephone calls over the Internet using a general purpose computer, rather than using dedicated hardware. A soft phone is usually used with a headset connected to the sound card of the PC. To communicate, both end-points must have the same communication protocol and at least one common audio code. Many service providers use the Session Initiation Protocol (SIP) standardized by the Internet Engineering Task Force (IETF)*
Keywords: *SIP; RTP; VOIP; RTCP; RTSP; IETF.*

I. INTRODUCTION

A soft phone is a software program for making telephone calls over the Internet using a general purpose computer, rather than using dedicated hardware. Often a soft phone is designed to behave like a traditional telephone, sometimes appearing as an image of a phone, with a display panel and buttons with which the user can interact. A soft phone is usually used with a headset connected to the sound card of the PC, or with aUSP phone.

The Soft Phone is an advanced, state-of-the-art communications application for your laptop or desktop PC. Enabled by AVVID (Architecture for Voice, Video and Integrated Data), the Soft Phone is a powerful voice-over- (VoIP) software application that is fully integrated with the Systems line of IP telephones. soft phones are software programs that can be installed on a computer or laptop. Implement the extension numbers (like 301, 302, 303….) just like how IP Hard phones are. So, calls can be transferred to these Soft phones using their extension numbers, and if DID is available, outside callers can directly call the Soft phone extension numbers. Soft phones have software based visual dial-pads on the monitor which can be used (by clicking on them using a mouse) to dial out/ receive calls, and for normal telephony functions like caller ID display, call hold, message waiting, call history, redial, etc.

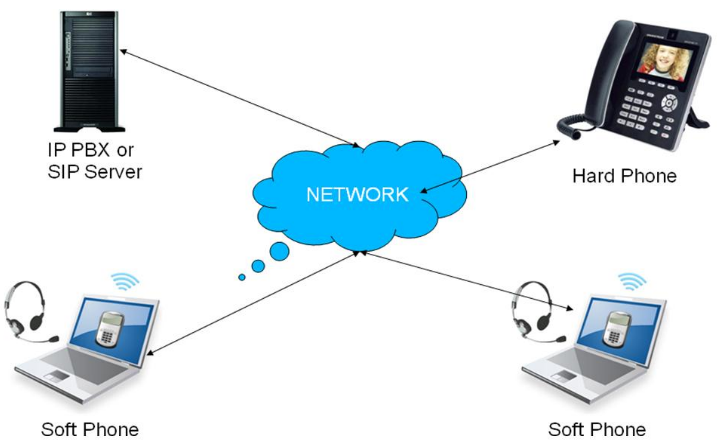

Fig 1 Soft phone Block Diagram

A. Advantages of SOFT PHONE are as follows:

- Soft phone uses existing Internet bandwidth and does not require any dedicated connection for voice communication Communication link lives for a short period of time.
- Since it does not require any dedicated channel, it uses bandwidth efficiently.
- Video calling and multi-party conferencing can also be done using Soft phone without incurring any additional cost.
- Easy to add new line without doing any additional wiring.
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B. SOFT PHONE Requirements:

a) Functional Requirements: The context of the system refers to the various connections and relationships between the system and its environment. The proposed architecture shows that the communication between the two soft phones by using SIPS and RTP protocols.

b) User Requirements: It provides requirement of he ystem, user or business, taking into account all major classes/categories of users. User requirement information can be in text or process flow format for each major user class that shows what inputs will initiate the system functions, system interactions, and what outputs are expected to be generated by the system. In the proposed system the input should be given in the appropriate field and the communication is established between the two nodes (softphones*)*.

c) Non-Functional Requirements: The requirements determine available resources required, response time, transaction rates, throughput, benchmark specifications and everything that deals with the performance of the system.

II. TYPES OF SOFT PHONES PROTOCOLS

Soft phone faces several protocols [8], they are as follows;

A. Session Initiation Protocol (SIP)

The Session Initiation Protocol (SIP) works in concert with these protocols by enabling Internet endpoints (called user agents) to discover one another and to agree on a characterization of a session they would like to share. For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests. SIP is an agile, general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols and without dependency on the type of session that is being established.

B. The Real-time Transport Protocol (RTP)

The Real-time Transport Protocol (RTP) defines a standardized packet format for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications and web-based push-to-talk features.RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video), RTCP is used to monitor transmission statistics and quality of service (QoS) and aids synchronization of multiple streams. When protocols are used in conjunction, RTP is originated and received on even port numbers and the associated RTCP communication uses the next higher odd port number.

C. Voice Over Internet Protocol (VOIP)

Is any of a family of methodologies, communication protocols, and transmission technologies for delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms frequently encountered and often used synonymously with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, and broadband phone.

D. Internet Protocol Private branch exchange(IPPBX)

An IP (Internet Protocol) PBX (Private branch exchange) is a business telephone system designed to deliver voice or video over a data network and interoperate with the normal Public Switched Telephone Network (PSTN). VoIP (Voice over Internet Protocol) gateways can be combined with traditional PBX functionality enabling businesses to use their managed intranet to help reduce long distance expenses, enjoy the benefits of a single network for voice and data and advanced CTI features or be used on a pure IP system which in most cases give greater cost savings, greater mobility, and increased redundancy. An IP-PBX can exist as a hardware object, or virtually, as a software system.

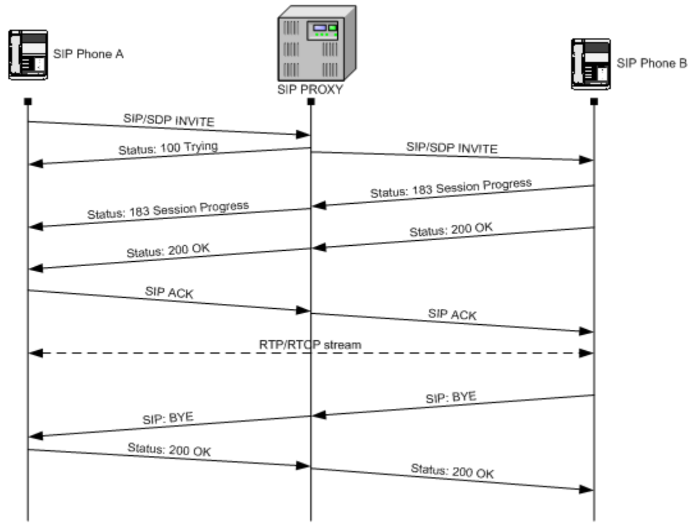

Fig 2 Overall Architecture

III. EXISTING WORKS

Most of researchers have focused on solving the overload problem in core servers, and therefore overload in edge servers still exists. Over the past decade, voice communications have moved from the public switched telephone networks to the Internet. The IETF has enabled this transition by developing stable standards for call setup and media transport. In fact in heavy overload conditions the server must devote all its resources to the rejecting requests so that its throughput will be virtually zero.

Two of the major advantages of soft phones (even some free ones) are instant messaging & video calling. But only the users who use similar soft phones can use these features. Some vendors even show presence information (availability information) of the contacts on IM that is built into the soft phones & some vendors support H.264 video compression standards for video calling to get best video quality, with least bandwidth. Soft phones can call IP hard phones that are directly or indirectly (through a service provider) connected to an VOIP/ IP Telephony Server as Soft phones are like any other extension of the IP Telephony server.

The process is also known as feasibility study**.** The development team visits the customer and studies their system. They investigate the need for possible software automation in the given system. By the end of the feasibility study, the team furnishes a document that holds the different specific recommendations for the candidate system. It also includes the personnel assignments, costs, project schedule, target dates etc. The requirement gathering process is intensified and focused specially on software. To understand the nature of the program(s) to be built, the system engineer or “Analyst” must understand the information domain for the software, as well as required function, behavior, performance and interfacing.

By using Soft phones, companies can not only save on the IP Hard phone costs, but they also save on cabling/ switch port/ electrical power costs. Depending on the vendor, some of them support a lot of additional features required in an enterprise like 802.1p Quos packet tagging (so as to prioritize voice traffic on the network), recording the voice conversations. Since soft phones are software programs, they are operating system dependent. They pop up on the screen when an incoming call is coming, so that the user could be informed on the same and pick up the call. Since most of the users these days work on their desktop computers for doing almost all of their work, Soft phones can be considered as a replacement to hard phones for many.

IV. PROPOSED MODEL

In the proposed system, the behavior of SIP proxy server under overloads and provides a better communication between two entities or soft phones. It is shown that this SIP proxy server performs better in term of throughput than a commonly used SIP rejection mechanism, which is response for management of overload condition.

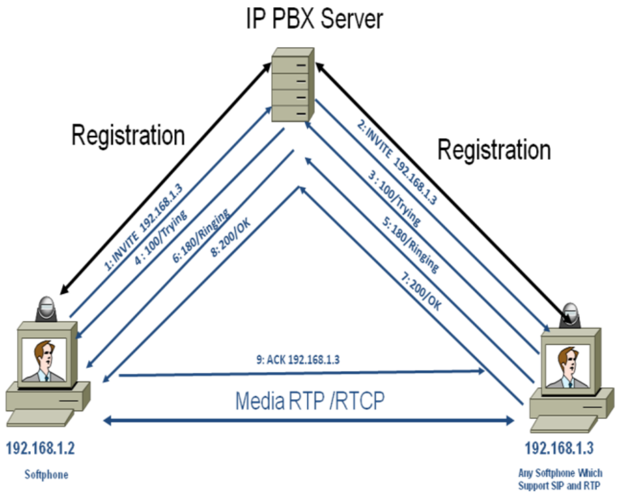

Fig 3. Proposed Model of Soft phone

A. *Implementation Of Ip Pbx Proxy Module*

Implementation of IPBX the teksip application is to be installed in a server and then need to configure the teksip application with domain IP and sip port number. Then start the teksip application service and wait for the user agent. Install the teksip application in a server and configure the ip address as 192.168.1.3 then install multimedia soft phone application in same server and configure sip domain ip address 192.168.1.3 and then install the multimedia softphone in another laptop in same network and configure the local ip address 192.168.1.4 and server ip address as 192.168.1.3 and start the teksip then start the client application. Run TekSIP Manager from Start Menu / Program Files / TekSIP. TekSIP automatically configures itself at first run. TekSIP selects first available IPv4 address to listen and make a reverse lookup of that IPv4 address to obtain SIP domain information. If TekSIP cannot resolve selected IP address to an alphanumeric FQDN address, selected IPv4 address is used as SIP domain. TekSIP also checks if it is installed behind an UPnP supported NAT gateway. If so, TekSIP automatically detects external IP and display it on status bar. TekSIP also adds a reverse mapping for incoming UDP connections automatically (Default UDP port 5060).

a) *Algorithm*

1. Begin
2. If (mode == NORMAL)
3. Vi = Vi-1+ (d0^2- di^2) / n;
4. If ((di-1> threshold * di) && (di< head * pk))
5. sign_gen = -1;
6. end if.
7. else
8. sign_gen = 1;
9. di= di-1+ √Vi* sign_gen;
10. else
11. if (mode == SPIKE)
12. end if
13. if (di^k<= tail * old_d)
14. mode = NORMAL;
15. else
16. if (di^k> head* pk)
17. mode = SPIKE;
18. old_d = pk;
19. end if
20. end

B. *Proposed Model of Registration Module*

Registration binds a particular device Contact URI with a SIP user. Address of Record (AOR). A REGISTER message containing Address of Record sip:jan@iptel.org and contact address sip:jan@1.2.3.4:5060 where 1.2.3.4 is IP address of the phone, is sent to the registrar. The registrar extracts this information and stores it into the location database. If all process is succeeded then the registrar sends a 200 OK response to the phone and the process of registration is finished.

C. User Interface Module

In this module all the user actions are performed such as dial, answer, reject, hang on .A user can wants to communicate with other means he dial the corresponding number by using the dial pad provided. If it is a registered user the call can be forwarded to the corresponding client otherwise communication cannot be established. Whenever a call is forwarded to client, he can take appropriate actions like answer, reject and divert call to other client. If client wants to answer another call means current call can be hanged on and attend the new call .After finished the call an old call can be retrieved.

TABLE I. NOTATIONS

Notations	**Description**
SIP	Session Initiation Protocol
VOIP	Voice Over Internet Protocol
RTP	Real Time Transport Protocol
IETF	Internet Engineering Task Force
IP PBX	Internet Protocol Private Branch Exchange
PSTN	Public Switched Telephone Network

Fig 4. Tables

The above table shows the notations used in this paper.

V. CONCLUSION AND FUTURE WORKS

Should Have Easy User Interface that many technological innovations rely upon User Interface Design to elevate their technical complexity to a usable product. Technology alone may not win user acceptance and subsequent marketability. Should Support Standard Protocol for call handling (SIP) There are many applications of the Internet that require the creation and management of a session, where a session is considered an exchange of data between an association of participants. Should Support Inbound Call a call that you are receiving on your soft phone. Should Support Outbound Call a call that you are dialed out to another soft phone. Should Support Audio Streaming (Input From MIC and Output in Speaker).

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