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A Proposed Solution for Web Conferencing System

[1]Snehal Sherkhane, [2]Sachin Bojewar
[1]Post graduate student, [2]Assistant Professor,
[1]Department of Computer Engineering,
[1]Shree L.R. Tiwari College of Engineering, Thane, India

***Abstract* - In today's scenario, with so many emerging technologies have been developed rapidly with the increase in use of internet users there is always a requirement for smart, easy to use and effective applications to meet various business and individual needs. Multimedia conferencing always plays a vital role for the purpose of interactive communication over a network. The most important factor to consider here is the protocol used for communication. H.32x based conferencing has already been used and widely deployed in the market for the past few years. SIP, i.e. Session Initiation Protocol, which is more viable and practical, is put on the agenda of being a call signaling protocol for conferencing. In this paper, we had a proposed solution by replacing the traditional H.323 protocol by SIP protocol for web conferencing. We will see the advantages of using SIP based protocol over H.323. We intend to implement the application with the help of web services using WCF i.e. Windows Communication Foundation (.NET Framework 4.0). The aim of this proposed solution is to provide a user friendly conferencing environment for reliable communication to the end users.**

***Index Terms* - H.32x, RTP, SIP, Web Services.**

I. INTRODUCTION

Multimedia Conferencing is one of the most fundamental aspects of communication. It has been on the research agenda for many years and is rapidly gaining attention in the field of communication. There are already few products which are based on H.32x standard. SIP i.e. Session Initiation Protocol, which is a more feasible protocol, is put on the agenda of being the call signaling protocol for the purpose of web conferencing. Most of the researches based on SIP conferencing have still remained unimplemented [1].

The various applications of conferencing have been developed and therefore are very advantageous to people who don't want to spend their time and money flying all over the world for face-to-face meetings. The International Telecommunication Union (ITU) has made H.32x standard, and the Internet Engineering Task Force (IETF) carries out SIP [2].
When we evaluate both the H.32x and SIP protocol, it has been found that H.32x is extensively deployed and more mature only because of its early adoption by the market, but has several problems, which includes scalability issues. On the contrary, SIP is more lightweight, flexible and extensible [3].

Besides protocol we need to consider other factors for implementing the application. Traditionally, a chat application uses some TCP socket based code that will maintain the socket between server and the clients. The major drawback of this technique is that on the server side a thread of pool is been holed back. Also keeping the performance aspect in mind, it can be said that this method may not extent very well. Instead of using TCP socket based code, web services can be used to commune events or messages among the request and the response. The major benefit of using a web service is that you don't have to manage the socket yourself or worry about configuring firewall on every client device to open a port for you, HTTP runs on port 80, and it is not blocked [4]. Hence, it is planned to develop a real time messaging application in .NET which will be based on HTTP long polling.

The rest of the paper is structured as follows. In Section II, we will briefly observe the protocol used in the existing system such as H.32x based, as well as for the proposed system such as RTP and SIP, see the benefit of using SIP over H.32x. In Section III, we enumerate the web services used. In Section IV, we will describe briefly the design consideration and prototype of implementation for our proposed solution. Finally, the conclusion and outline of future research will be presented in Section V.

II. PROTOCOLS

The following section will briefly gives us information of all the protocols which are essential for the application which consist of H.32x, RTP and SIP protocol.

H.32x Standard Protocol

It might cause a problem while handling all the data packets related to conferencing, as the users work with different environment. This calls for a standard to which the software must be conventional. To aid the compatibility among diverse vendor solutions, ITU i.e. International Telecommunication Union, developed H.32x standard series; including H.322, H.323, H.324. H.322 is for applications over LANs providing a guaranteed bandwidth. H.323 is for applications over LANs, providing non-guaranteed bandwidth and H.324 is for applications over the PSTN and mobile phones. Among them H.323 standard enables networking and application vendors to support more deployable, manageable and affordable conferencing as well as it enables interoperability and compatibility.

Real- Time Transport Protocol

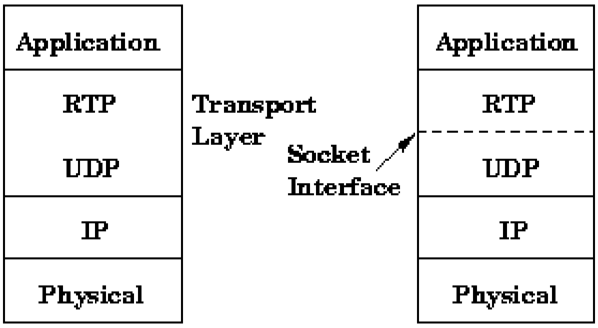

Fig. 1 Location of RTP in IP stack

It is a transport protocol for real-time applications and used for real-time data transfer which includes audio and video. RTP cannot guarantee us with a quality-of-service functionality for the real time services. Various properties such as loss detection, security, content identification, and continuous media are supported by RTP for real time applications. RTCP (Real Time Control Protocol) primary purpose is to provide a feedback on the quality-of-service in the media distribution by occasionally sending the entire statistics information to users in a streaming multimedia session. Both the RTP and RTCP are designed in a way to have an independent existence of the underlying transport and network layers [8].

Session Initiation Protocol

There are various applications for conferencing system and this entire system framework uses different methods. Here H.323 based conferencing is already been used for quite some years, but SIP- based on the other hand is still in emerging stage.

SIP has been developed by International Engineering Task Force i.e. IETF multimedia session control working group. It is an application layer protocol that can establish a session, modify a session, and terminate multimedia sessions. It is said be a text-based protocol and uses UTF-8 char set. It can be integrated with other IETF protocols to build a complete set of multimedia architecture. There are six basic methods such as REGISTER, INVITE, ACK, CANCEL and OPTIONS [9].

SIP over H.32x

Transmitting media data across the network in real-time requires high throughput. It's better to compensate for data loss than to compensate for large delays. When extended to world-wide networks, SIP has many benefits: [5]

Loop detection: A loop can occur when trying to trace a user over numerous domains. There is no support for loop detection in H.32x. With the help of SIP header, loops are easily detected, as they specify all the required proxies handling the SIP packet.

Distributed control: In H.32x, when we perform multi-user calls we need a central point such that it should be dimensioned as per the size of the call. The SIP session is dispersed, therefore the need for central point is vanished.

Small connection overhead: It takes almost three times the data and turnarounds in H.32x compared to SIP while establishing a connection.

Some other advantages include: [6]

Simplicity: It includes only six methods which makes it simpler compared to other protocols.

Independence of the transport layer: It can be used in conjunction with other protocols such as ATM (Asynchronous Transfer Mode), TCP (Transmission Control Protocol), UDP (User Datagram Protocol), etc.

Personal mobility of users: Due to the task of having a unique identifier for all the users they can therefore move anyplace within a network without any limitations.

Network Scalability: The overall structure of SIP protocol makes it quite easy to expand as well as increase its number of components whenever required.

Extensibility: The protocol is categorized by the prospect to add-on with new features whenever new services emerge.

Integration with existing IP stacks: The protocol is part of a global multimedia architecture which is developed by the IETF (Internet Engineering Task Force) which includes an RSVP (Resource Reservation Protocol), RTSP (Real Time Streaming Protocol), RTP (Real Time Transport Protocol), and SDP (Session Description Protocol).

Interaction with other signaling protocol: It can be used in combination with various IP telephony protocols, PSTN protocols, and also used to communicate with a smart grid.

III. WEB SERVICES

The conferencing applications can be developed using web services either by using the principles on SOAP or on REST, which are briefly described below [7]:

SOAP (Simple Object Access Protocol), consist of three entities which include: a service provider, which will publish the required service in the registry; a service registry, which acquaint us with the online service discovery and a service requestor, which will find all the services by querying the registry. Here, messages are exchanged over HTTP easily. Web Service Description Language provides the necessary information of web services.

REST (Representational State Transfer), uses the traditional client- server architecture, which is mostly used with HTTP but does not confine itself to a particular protocol. It is described using the Web Application Description Language, the file here describe the request which can be addressed for a service also including the service's URI including the data the service expects and the servers. REST depends on three main design principles: addressability, uniform interface and statelessness.

IV. Proposed Solution

Our proposed solution is a plan to implement a web application called as Scalable Web Conferencing System Using WCF (Windows Communication Foundation) framework which can be used for audio and video chat and which will be more scalable, web services are used for the purpose of communication over the network and SIP protocol is used instead of the H.32x for call signaling and various other purposes. The following section will illustrate the design consideration and implementation plans for the same.

Design Consideration

Because of the advantages of SIP over H.323 discussed till now, SIP is selected as call signaling protocol. In this section, design characteristics of the conferencing system are listed below [1]:

1. Because SIP relates to HTTP it logically will have all the security mechanisms similar to HTTP. Verification of the caller and the callee can be realized with HTTP mechanisms, that includes basic authentication.
2. Currently, firewalls protect almost all the intranets. The cooperation of firewall is must for the SIP conference to take place. A proxy server is needed and it operates like most other proxies. Not only the firewall acts as a gateway, but also it decides which monitored calls are to be passed through. A proxy can be well thought-of as a gateway to control bandwidth along with the access to control policies.
3. SIP uses 5060/5061 port addresses. Port number 5060 for the SIP connection is assigned, when the call is set- up and a new connection is established. Two UDP connections are required by each media channel for the RTP streams, wherein for the RTCP streams one bi-directional connection is required. For audio-only conference we required two TCP connections and four UDP connections and only one among them is static.
4. The proxy has to contribute with the application protocol making the firewall detectable to the applications. The firewall looks like a server to the SIP conferencing system, whereas for the external application the firewall looks like a client. This kind of proxy can be classified as an application proxy. In SIP, the firewall does not cause that much problems. Only one UDP or TCP connection is needed and it is easy to add to the firewall configuration.
5. This multimedia conferencing system does not only support audio or video like traditional products. It can definitely be called a data conferencing system, as it also enables document and image exchanges among multiple participants.

Implementation Plan

With the help of gathering design consideration explained in above section we can get a brief idea on the implementation part of our project. Here .Net Framework 4.0 will be used owing to the fact that it provides an easier way to create application that present, capture, manipulate, and store time- based media. In .Net, C# will be used to reduce the code, making it simple as well as reduce the overall size of the application. The database which will be used is in SQL. Web Services will be used instead of TCP socket based code. We will implement the application as Silverlight + Polling web services, which will share the information between users over HTTP in real time.

The SIP Conferencing Architecture as shown (Figure 2) consists of a main conference server and a single participant/ user. Here, a focus which is a SIP user agent is in charge for the management of conference using SIP signaling protocol. The conference policy contains all the necessary rules and guidelines that will help in making the decision process of the focus for the management of all the requests from the participants during conferencing, it can also act as a notifier by accepting the required subscriptions to the state of the conference and also to notify the change in conference state whenever essential.

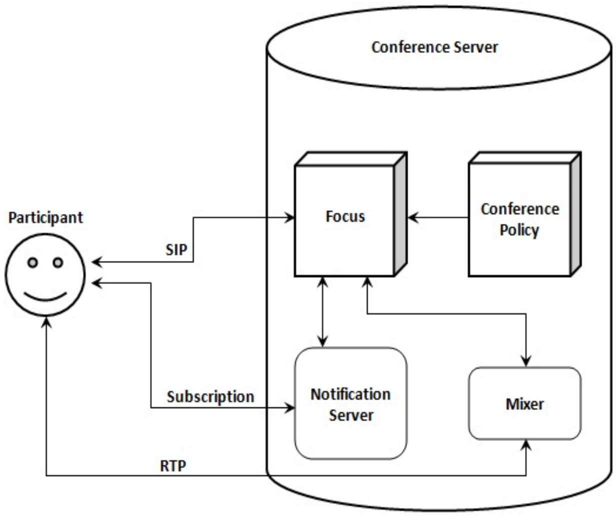

Fig. 2 SIP Conferencing Architecture

The notification server, is nothing but the logical function provided by the focus. The mixer is controlled by the focus and is liable for handling the multimedia streams, and generating output streams which further can be distributed to the users [9].

The above system which is implemented for a single participant can be improvised by having multiple users and therefore implementing a multi user conferencing system.

V. Conclusion

Video Conferencing has helped people to communicate effectively. The early conferencing solutions are based on H.32x standard. H.32x has reached a good position for the purpose of communication protocol but when advanced, flexible and scalable conferencing are needed, SIP is found to be a good option. Therefore the various advantages of SIP such as simple, scalable, extensible, and human readable, etc. makes it a better choice for our application to be implemented in. This project will be an effort to eliminate incompatibility between different platform- oriented solutions, provide more users friendly and scalable conferencing environment and adding more features for the end user.

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