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A Web Conferencing System Using Wcf Framework

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***Abstract*—In this upcoming phase of so many numerous and countless promising technologies catching up and also the day to day challenges which they cause to face requires constant progress and up-to-date by introducing most simple, smart, effective and easy to use applications for the end users to meet various business as well as individual needs. The Multimedia Conferencing always plays a vital role for an interactive communication over a network. Owing to the fact that a conferencing system requires real time properties, we will consider all the necessary factors to make the system more vibrant and efficient. In this paper, we intend to implement a web conferencing system with the help of web services using WCF, i.e. Windows Communication foundation (.NET Framework 4.0) which provides an easier way to create applications that capture, present, manipulate and store real-time based media. This application will incorporate all the real- time services such as video chat, audio messages, file manager services, whiteboard services in one single framework with the help of Silver light, a plug- in power of .Net Framework. The purpose of this report is to offer a user friendly conferencing environment for reliable communication to the end users.**

Keywords—H.32x; RTP; SIP; WCF; Web Services

I. Introduction

One-to-one interaction or communication has gained popularity today with the support and background of so many applications in use and still developing but when it comes to one-to-many or many-to-many interaction there is still a lot of scope and job to be done to make it more powerful and efficient. The Multimedia Conferencing is the most elemental aspects of communication in today's scenario where we bear so many challenges and task to perform and handle. The conferencing system has constantly been on the research agenda for quite some years and has acquired interest in the area of communication.

Previously for the purpose of communication socket programming was preferred and handled by writing some TCP socket based code that sustains sockets between different server and the clients. The client initially needs to request the server, a socket is open, the server processes the request, results get generated, and further socket gets closed after process completion. Also keeping the performance aspect in mind, this technique may not scale very well. If the application is served by a keen server with a prearranged amount of clients, this solution can be still persuasive. Instead of using TCP socket based code, web services can be used to interconnect events or messages between the invitation and the answer. The benefit of using a web service is that you don't have to manage the sockets yourself. You also don't have to worry about constituting every firewall on every client device to open a port for you. HTTP runs on port 80, and it is not blocked. Your clients can link with each other over HTTP, without firewalls. [1]

In multimedia networks, maintaining the precise QOS level is fundamental. The video conferencing can impose strict requirements and arise various factors that may alter the quality of service in the network which we also need to consider. [2]

Another important factor which we need to consider is the protocol to be used for the purpose of the call signaling during conferencing. There are already few products which are based on H.32x protocol standards for communication. SIP, i.e. Session Initiation Protocol, which is comparatively more viable protocol, is put on the schema of being the call signaling protocol for conferencing. Most of the study and researches on SIP based conferencing, however, have still remained unimplemented. [3]

The rest of the paper is structured as follows. In Section 2, existing work related to this system has been highlighted. In Section 3, protocol related to the system has been described briefly. In Section 4, the proposed framework for the system has been explained thoroughly. In Section 5, screenshots, related to the application has been shown. In Section 6, comparison has been done by well known similar applications. Finally, the last Section consists of conclusions and future work related to the system.

II. Related Work

In this Section, we will run through all the existing work carried out which are related to the application we want to implement. Here we will briefly introduce the WCF concept; the existing protocols used for conferencing, and also briefly emphasize the web services.

A. WCF

Windows Communication Foundation (WCF) is a structure for building service-oriented applications. We can direct data as asynchronous messages from one service endpoint to another. A service endpoint can be a portion of a continuously accessible service hosted by IIS, or it can also be a service hosted in some application. An endpoint called as a client of a service requesting data from a service endpoint. The messages here can be as modest as a single character or word sent as XML, or as intricate as a stream of binary data. WCF is a malleable platform. Because of this advantage, WCF is also used in numerous other Microsoft products. Microsoft Silverlight is a platform for forming interoperable, rich Web applications that allow makers to create media-intensive Web sites. Inauguration with version 2, Silverlight has united WCF as a communication technology to associate Silverlight applications to WCF endpoints.

There are also many features of WCF which makes it the best choice for us to implement our application using this framework. It includes features such as: Service orientation, Interoperability, Data contracts, Reliable Messages, Multiple Encodings and Security support. [10]

Also, there are several advantages of using the WCF framework such as: [11]

- It is interoperable with other services
- Its services provide improved consistency and security in contrast to ASMX web services.
- The security model here does not need to make any changes in the code and binding properties. Just a few changes in the configuration will make your requirements.
- It has incorporated logging mechanism, altering the configuration file settings will provide this functionality. In other expertise developer has to write the code.
- Load balancing and also supports scalability.

B. Existing Protocol: H.32x Standard

It may cause a problem while managing all the data packets related to conferencing, as the users work with different location and environment. This calls for a standard to which the software must be convened. 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 telephones. 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. [4]

C. 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 [5]:

SOAP (Simple Object Access Protocol), consist of three entities which include: a service provider, which will issue the required service in the registry; a service registry, which introduce us with the online service discovery and a service requestor, which will ascertain all the services by querying the registry. Here, messages are exchanged over HTTP easily.

REST (Representational State Transfer), employs the traditional node- server architecture, which is mostly used with HTTP but does not limit itself to a particular protocol. REST depends on three primary design principles: addressability, uniform interface and statelessness.

III. Protocol Description

Here we will study the protocols required in our application more specifically and in detail. The basic Real- time Transport Protocol and Session Initiation Protocol has been described below.

A. Basic Real- Time Transport Protocol

It is a transport protocol for real-time applications and used for real-time data transfer, which includes audio and video. Various properties such as loss detection, security, content designation, and continuous media are supported by RTP for real time applications. RTCP (Real Time Control Protocol) primary aim 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 manner to sustain an independent existence of the underlying transport and network layers [6].

B. Session Initiation Protocol

SIP is a lightweight, text-based signaling protocol used for establishing sessions in a network. SIP is an application-independent protocol; it simply initiates, terminates and modifies sessions without knowing any details of the sessions. This simplicity means that SIP was designed at the outset to be extremely flexible, scalable and extensible. It is a text-based protocol which can easily interact with other internet protocols. SIP is gradually becoming popular because of its excellent characteristics. [3]

SIP is not a perpendicularly integrated communications system. It is rather a module that can be used with other IETF protocols (e.g. RTP, RTSP and SDP) to build a complete multimedia architecture. [7]

There are five services for establishing and terminating a session in SIP [8]:

User location: determining the goal system to be utilized for communication;

User availability: determining inclination of the called party to engage in communications;

User capabilities: determining the media equally well as the parameters to be applied;

Session setup: "ringing", establishing of session parameters at both called and calling party;

Session management: transfer and closure of sessions, adapting session parameters, and summoning services.

When adding the functionality of scalable and more reliable protocol services for call signaling the existing protocol used for conferencing, which is H.32x standard has few limitations when compared to SIP i.e. Session Initiation Protocol.

Following are the advantages of SIP over H.32x standard: [9]

- Loop detection: When trying to localize a user over several domains, loops can occur. H.32x has no support for loop detection. Loops are certainly detected using SIP headers, as they specify all proxies that have handled the SIP packet.
- Distributed control: In H.32x, there is a need for a vital point when performing multi-user calls, which means that this vital point must be dimensioned for the size of the call. SIP sessions are scattered, making the need of this central point disappear.
- Small connection overhead: Launching a connection using H.32x takes about three times the data and about-turns paralleled to when using SIP.

IV. PROPOSED WCF ARCHITECTURE

The following figure shows the proposed model of our application in which WCF framework has incorporated major portion of conferencing system in one single framework.

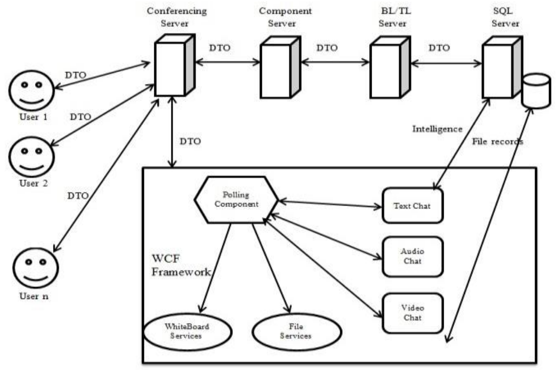

Fig. 1. Proposed WCF Architecture Model

The above proposed model has merged all the services in one complete framework called as Windows Communication Foundation. The framework is more scalable and reliable because it uses the concept of Silverlight tool and replaces the traditional H.32x standard protocol with Session Initiation Protocol. Here in this model we have many different components and each component is associated with some functionality. The multiple users/ participants are acting and participating in this system. The Component Server or Controller, consist all the relevant messages which have methods and protocol information, a number of header fields which specify the call properties as well as the service information and an optional message which consists data relevant for the session description.

The Conferencing server is basically responsible for managing the entire conferencing system with the help of a SIP signaling protocol. Managing services include: authentication, authorization and communication security from eavesdropping and manipulation of the contents. It also acts as notified, accepting subscriptions to the conference state. The Business Layer/ Transport Layer has the same functions of guiding the decision making process. The WCF framework has merged all the main services of the conferencing system in one single picture. Component is responsible for controlling the two main services called as whiteboard services and file services. Both these services can act independent services or can be merged in one single component. Other small components such as text chat, video chat and audio chat have an important role to play during conferencing and thus needs to be merged and linked effectively. Other components such as SQL server consisting all the data and important information such as file shares during the conferencing, their entries, etc. on this component.

The DTO i.e. Data Transfer Object is a design pattern used to transfer data between different software application subsystems.

V. SCREENSHOTS

1. Login Screen

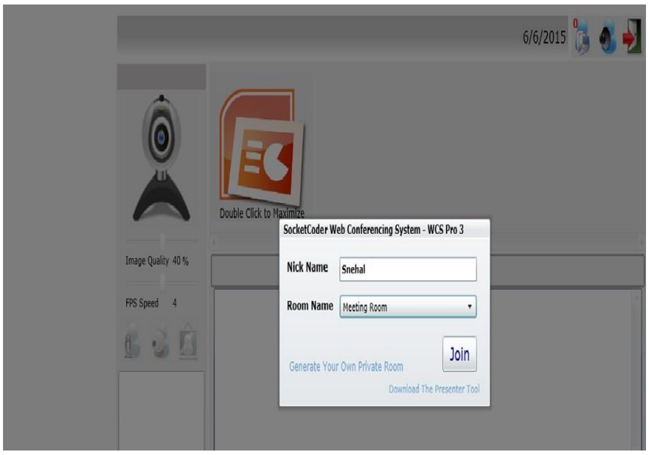

Fig. 2. Login Screen

2. Successful login of participant 1

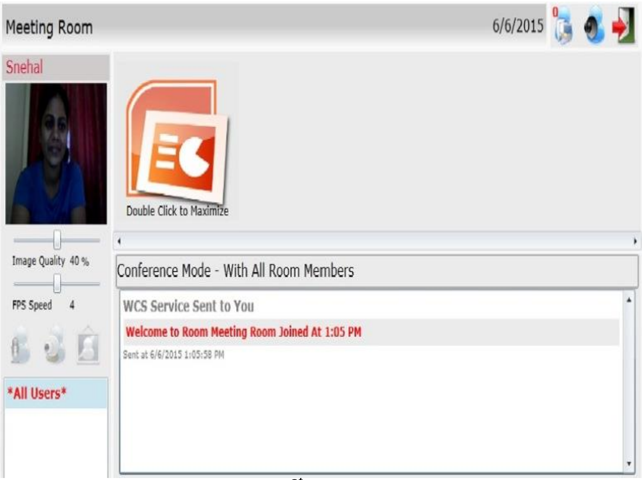

Fig. 3. Successful login of 1st participant

3. Successful login of participant 2

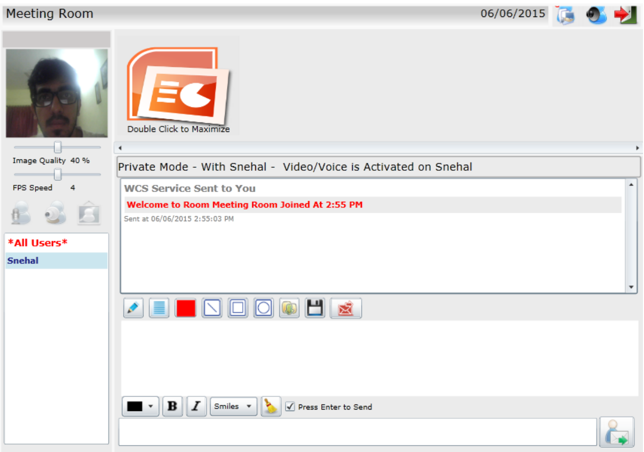

Fig. 4. Successful login of 2nd participant

4. Participant 1 and 2 texting and video chatting among themselves

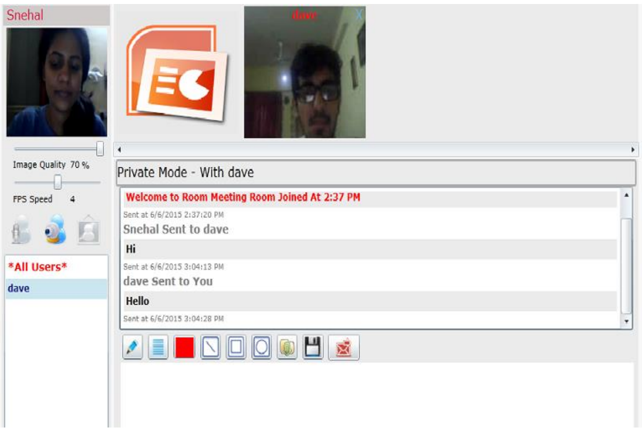

Fig. 5. Participant 1 and 2 texting and video chatting

5. Participant 2 using the whiteboard seen at participant 1's end

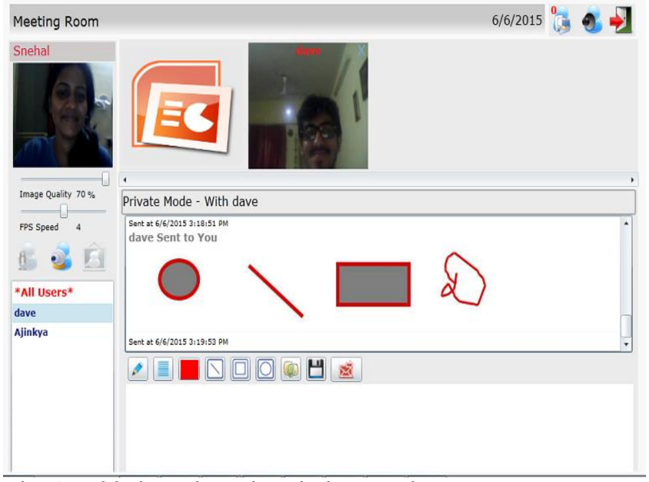

Fig. 6. Whiteboard services being used among users

6. Participant 1 sharing a file

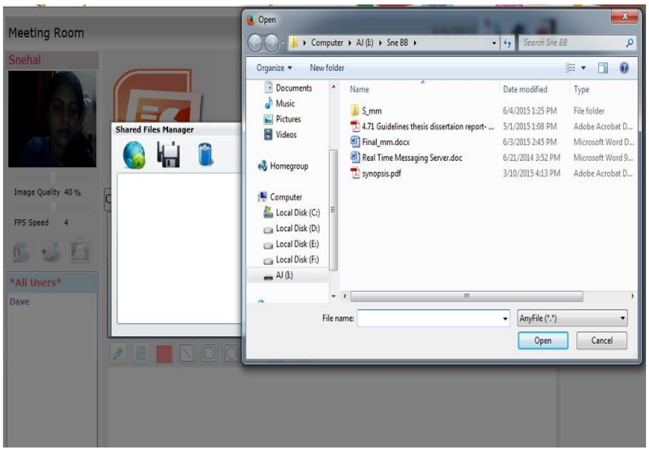

Fig. 7. Participant 1 sharing a file

7. Participant 2 saving a shared file

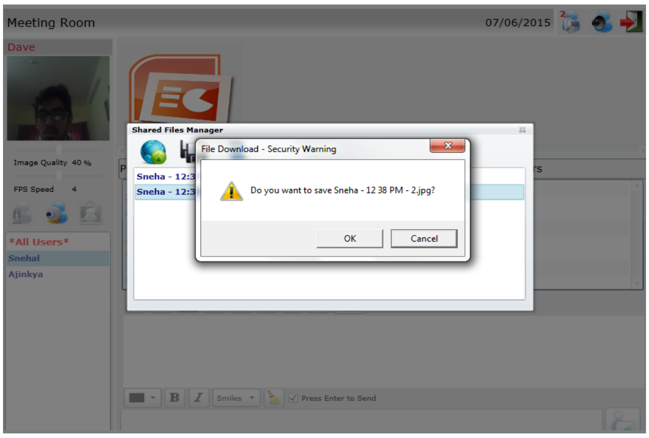

Fig. 8. Participant 2 saving a file

Other then the screenshots showed there are many other features of this application such as various file sharing properties such as: removing, sharing on private mode. Many white board services has been captured and implemented successfully. The main services of audio and video conferencing has also been implemented. Here user can also have a private mode chat during conferencing with others if he feel to need some private information only with a single user. Many other functionality has been implemented successfully for a simple and effective multimedia conferencing system.

VI. Results And Discussion

The comparison is built on a WCF/ SIP based system with some other well know conferencing application. The tested environment is between 4 users. With the help of bandwidth monitoring tool we have recorded the performance with respect to the consumed bandwidth.

MSN:
Download speed: 9.15 kbps
Upload speed: 8.38 kbps

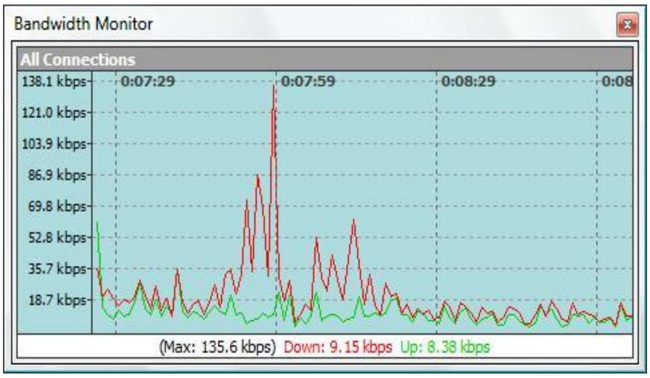

Fig. 9. Bandwidth consumption by MSN

YAHOO:
Download speed: 16.0 kbps
Upload speed: 8.40 kbps

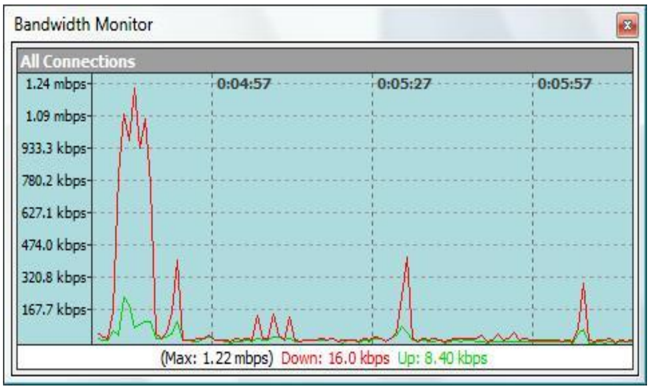

Fig. 10. Bandwidth consumption by YAHOO

GMAIL
Download speed: 20.0 kbps
Upload speed: 16.5 kbps

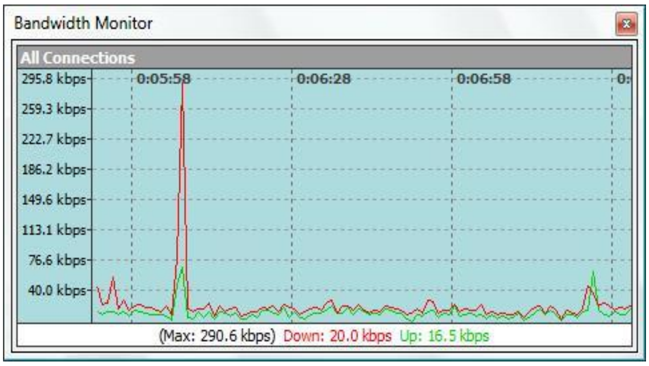

Fig. 11. Bandwidth consumption by GMAIL

SKYPE
Download speed: 31.9 kbps
Upload speed: 10.6 kbps

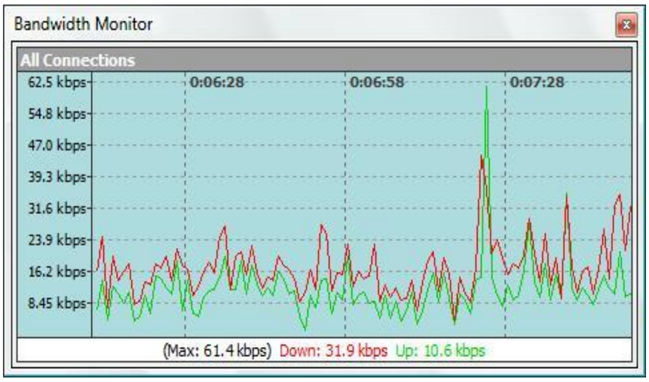

Fig. 12. Bandwidth consumption by SKYPE

WCF Based system:
Download speed: 228.5 kbps
Upload speed: 32.2 kbps

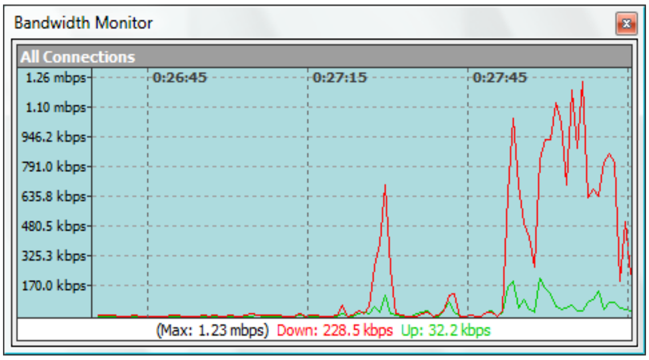

Fig. 13. Bandwidth consumption by WCF System

VII. Conclusion And Future Work

Video conferencing has helped people to communicate more effectively. Instead of using TCP socket based code web services have been used for the purpose of communication. The WCF framework has been deployed where Silverlight tool is used to incorporate the entire platform in one single picture and making the application more simple and user friendly. SIP protocol is used instead of the H.32x for call signaling and various other purposes. For interactive interface and user experience additional features such as whiteboard and file sharing has also been implemented. This paper is part of the effort that aims to eliminate incompatibility between different platform-oriented solutions, by providing more simple, user- friendly and scalable application for the end users.

In future studies a variety of new functions to meet advanced requirements, such as inviting 'n' number of users to the conference, dial-out method, etc. still needs to be implemented. Mining structures that will give a brief history of the user, topic of interest, etc., can be implemented as an additional intelligence feature.

REFERENCES

[1] Arun Singh, Ajay Kr. Sharma, Ashish Kumar, ' Multiclient Multi Instance Centralized File Server using TCP protocol in Java Sockets', International Journal of Computer Applications, June 2011.

[2] Catalina Mancas and Mihai Mocanu, 'QoS optimization in Congested Multimedia Networks', Proceedings of the IEEE Transactions, March 2013.

[3] Junchao Li, Weimin Lei, Xiuwv Zhang, ' Design and implementation of a SIP- Based Centralized Multimedia Conferencing System', International Conference on Communication Software and Networks, ICCSN' 09.

[4] G. ITU-T Rec. H.323,' Packet based Multimedia Communication System', Telecommunication Standardization Sector of ITU, July 2003.

[5] Fatna Belqasmi, Jagdeep Singh, Suhib Younis Bani Melhem, and Roch H. Glitho, ' SOAP- Based vs. RESTful Web Services', IEEE Computer Society, 2013.

[6] A. Niemi, ' Session Initiation Protocol (SIP) Extension for Event State Publication', RFC 3903, IETF, October 2004.

[7] Junchao Li, Weimin Lei and Haiyang Li, ' A Comparison of Multimedia Conferencing Framework', Journal of Networks, vol. 5, no. 6, June 2010.

[8] A. Roach,' Session Initiation Protocol (SIP) Specific Event Notification', RFC 3265, IETF, June 2002.

[9] K. Katrinis, S. Zurich, G. Parissidis and B. Pcattner, 'A Comparison of Frameworks for Multimedia Conferencing: SIP and H.323, 8th IASTED International Conference on Internet and Multimedia System and Applications, vol. 6, pp. 291- 300, 2004.

[10] https://msdn.microsoft.com/enus/library/ms731082%28v=vs.110%29.aspx

[11] http://www.wcftutorial.net/Introduction-to-WCF.aspx