Voice and Video over Wireless LAN

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Abstract- "Voice and Video over Wireless LAN" is concerned with establishing audio and video calls using wireless LAN. We are using client server model for achieving this purpose. A wireless router is used to establish the wireless LAN. Establishing wireless LAN is easier, cost effective and less time consuming than establishing wired LAN.

This system provides flexibility of operation. We are configuring one system having minimum of 1GB RAM as a server. Server is responsible for establishing and maintaining connection among clients. Any PC within the range of WLAN can be configured as client. Installation and maintenance of wired LAN is tedious and expensive. Comparatively installation of WLAN is simple and quicker. Maintenance required is also less. Comparatively it is easier to troubleshoot. Hence we propose a wireless system for audio and video calls within an organization.

Index Terms—WLAN, VOIP, SIP, RTP

INTRODUCTION

Voice over IP (VoIP) has presented a unique opportunity for enterprises. Convergence the merging of data networks and voice networks over a common IP infrastructure - can offer a dramatic reduction in the capital and operational expense of maintaining separate voice and data infrastructures. Beyond these cost savings, the ability to host voice and data on the same network can lead to improvements whereby data applications can leverage unique multimedia capabilities, while voice and real-time multimedia applications are able to take advantage of rich enterprise data features that can enhance communications in a manner that can reduce the need for costly face-to-face meetings. Additionally, convergence can lead to a unique synergy resulting in the development of new real-time applications. An important element that is missing from this equation is mobility. The transition of VoIP to the wireless space is an inevitable extension of this trend, since Voice and Video over WLAN extends the reach of a company's IP telephony and multimedia communication systems, enterprise work forces from the confines of their offices, and opens the door to a new generation of wireless converged network applications. [1]

Given its rapid end-user acceptance, it is not surprising that wireless LANs have come to the fore as a growing part of the enterprise communications landscape. Early issues such as security have been addressed and companies are now

systematically consolidating access points into wireless enterprise infrastructures.

Wireless networks were originally designed for the wireless transmission of data. Therefore, adding voice and video presents several challenges that must be resolved before voice over WLAN can supplant traditional wireless voice solutions: best-in-class voice quality, robust security embedded in the corporate security model, support for both on and off-site mobility, high availability, and low total cost of ownership (TCO).

PROBLEM DEFINITION

Currently there are technologies existing for transmitting voice over long distance. However they are quiet expensive. There are systems like Skype, GTalk which are useful for low cost communication. Skype for example allows free call to first fifty pay tariff to Skype. If we wish to have more than fifty contacts on the same identity we need to pay tariff to Skype. In case we don't wish to pay then we need to open new account with new identity. For companies second solution is not recommended. Also server of Skype, GTalk are not accessible to administrator. For using these services we need to have access to net connection. It could be a costly affair for small companies. Installation and maintenance of wired LAN is tedious and expensive. Comparatively installation of WLAN is simple and quicker. Maintenance required is also less. [1] Comparatively it is easier to troubleshoot. Hence we propose a wireless system for audio and video calls within an organization.

EXISTING SYSTEMS

Name	Private	File	Video	Voice
	Chat	Transfer	Call	Call
iptux	No	Yes	No	No
iChat	Yes	Yes	No	No
Outlook LAN messenger	Yes	Yes	Yes	No
eBuddy	Yes	Partial	No	Yes

GTalk	Yes	Yes	Third	Yes
			party	
			plug-in	
Xfire	Yes	Yes	No	Yes

Table 1: List of existing systems

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PROPOSED SYSTEM

There are five main modules in this system. They are:-

- Registration & authentication
- GUI module
- Audio call
- Video call

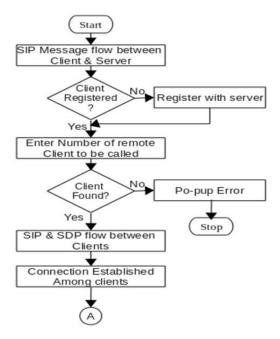


Figure 1: Registration and authentication

When we are running client for first time, we need to register client with server. Here we register our desired number with server. When a call is made to any client server it checks if referenced client is registered with it. If client is within range of wireless LAN then connection is established between both clients.

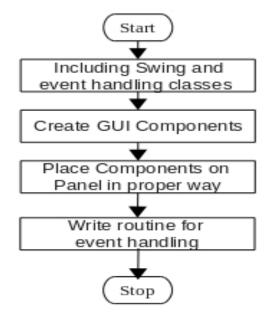


Figure 2: GUI Module

We need GUI for audio and video call. We made those GUI using swing. GUI includes GUI for registering with server, GUI for conferencing, GUI for call progress states. It also includes pop-up box for alerting user on arrival of call. The number of calling client and message is displayed using pop-up box.

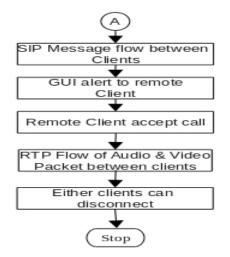


Figure 2: Audio and Video Call

We are using SIP for initiating and maintaining session. This module converts speech to bytes & then they are transmitted.DES encryption decryption algorithm for security of transferred data. Since audio, video call require various QOS parameters like delay & jitter within constraint we use RTP to transfer audio, video packets. For efficient transmission of video we need to compress it. Since bandwidth is important consideration in wireless network we are using MPEG compression for video transfer.

IMPLEMENTATION

Entire project is mainly based on two modules i.e. server software and client software. Each user must have the client software. Admin will have the server software. To run the client software server software must be in running condition to serve the request of each of the client. These programs can communicate using socket programming. Whenever we run the client software socket connection is established with the IP address of the server and the port number which is open for the communication.

Synchronization between client and server has been achieved with the help of the request messages i.e. client sends the "form no" to the server to tell the server which code the client is currently executing. Whereas server sends the unique field "Data sent=MESSAGE" to tell the client that there is message for you and similarly for other features like file sharing and calling it uses these types of messages.

Feature Implementation

A. CLIENT LOGIN

- Whenever client enters the user name and password on the home page and clicks on "sign in" button these fields are sent to the server side
- Server then checks whether the user is registered or not from the database and sends the reply accordingly
- As soon as the client gets logged in, server sends the profile information of the user and the list of online and offline users.

B. TEXT CHAT

- Whenever user enters a message it is stored on client side database (to maintain the history) and forwarded to the server
- Server also stores this message on the server database for offline users and forwards it to the receiver client
- On client side MS access is used and server uses 'rs2xml' API to take the data from the chat box and store it in the MS access file
- On the server side Oracle 10g is used as the database for which 'ojdbc6' API is used
- The receiver client again stores the message into the database and message is displayed on the screen

• On clicking on the particular user from the list chat history is loaded from the database in the chat box

C. OFFLINE MESSAGES

- User can also send the message to offline users, for this at the server side status of each message is maintained i.e. if the message is delivered then status is 'delivered' else status is 'pending'
- Whenever any user is logged in, server checks whether there is any pending message for that user
- If message is pending it simply delivers it to the client and updates the status of that message as 'delivered'

D. FILE SHARING

- Whenever client shares a file it is first sent to the server and then the server forwards it to the client
- The entire file is divided into the packets of size 1024 bytes and these packets are sent to the server in the form of a byte array
- In this way the entire file is transmitted to the server and in a similar way server transmits the file to the receiving client
- File is stored temporarily on the server so that if the client is offline it can be forwarded to him whenever he is available

E. VOICE CHAT

- To implement voice chat SIP protocol is used which uses special type of messages for establishing, maintaining and terminating the call
- In this system whenever client clicks on the call button the request is sent to the main server, server then forwards the request to the receiver
- Receiver then accepts/rejects the request, this reply is given back to the caller.
- For voice call we have used a special type of mechanism, in which audio data is first recorded and then forwarded to the receiver
- We have used two threads, one for listening to the caller and another for transmitting recorded voice
- To achieve this we use 'Multi chat 1.0' API

F. VIDEO CHAT

For establishing video call we are using classes in the JMF package.

Classes used in this Module:

1) AVReceive2

AVReceive2 performs the following tasks:

- Open one RTP session per session address given.
- Listen for the NewReceiveStreamEvent from the ReceiveStreamListener.
- Create a JMF Player for each stream received for playback.

2) AVTransmit2

This class is used to transmit audio and video on two different ports and the IP address of the receiver

3) Player

This class is used to start the web cam on both the sides

4) CaptureDevice

This class is used to detect audio and video capture device

5) Capture

This class is used to capture the video from web cam

GRAPHICAL USER INTERFACE



Figure 3: Login Screen



Figure 4: User Profile Screen



Figure 5: Conversation View

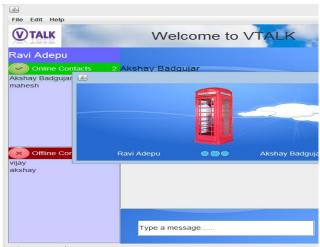


Figure 5: Call initiation screen

CONCLUSION

Currently systems are available in market for inexpensive communication. However they have constraints like need for internet connection. In this project we attempt to introduce cheaper audio and video communication over wireless LAN. We achieve this using WLAN based system. As it requires only wireless router, personal computer & does not require internet connection, this system is very cost effective. It is easy to set up the system as no additional wiring is required in case of conventional system used for communication. This system will be of great use in small scale industries as well as educational institutions. It will be helpful as cost cutting means that facilitate audio video communication.

FUTURE WORK

The motive of this paper was to propose a system for Simple Voice and Video transfer over WLAN.

This system establishes communication between only two participants. However, future developments may explore implementations for mobile platforms, video conferencing as well as desktop sharing.

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