Voice and Video over Wireless LAN

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Abstract:- **World is becoming a global village. As a result communication is gaining increased importance. The wireless LANs are becoming more & more popular because they can satisfy the requirement like mobility, relocation of user, ad-hoc networking & coverage of locations which are difficult to wire. Earlier the wireless LANs were costly, could support only low data rates, a license was required. Hence there were limitations on the practical utility of wireless LANS. But all these problems are being addressed now which is increasing the popularity of wireless LANs day by day.**

Our paper "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. D-Link wireless router is used to establish the wireless LAN. Establishing wireless LAN is easier, cost effective and less time consuming than establishing wired LAN.

Keywords:- **WLAN, TCP/IP, SIP, SDP, RTP**

1. 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.

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.

Providing real-time communication like voice or video over WLAN is a technically challenging task. The users of mobile devices expect at least the same functionality and performance they receive from traditional voice solutions. However, wireless networks were originally designed for the wireless transmission of data. Therefore, adding voice and video presents several challenges that must be resolved before VoWLAN 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).

2. RELATED WORK

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 contacts. If we wish to have more than fifty contacts on 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. Comparatively it is easier to troubleshoot. Hence we propose a wireless system for audio and video calls within an organization.

Wireless LAN offers an easy and cost effective alternative to existing model of communication. Their greatest advantage is they provide mobility to users. It is easy to setup WLAN in offices as it doesn't require laying cables to individual PC. With development in field of networking it is becoming cheaper to install wireless LAN. Development of low cost memories and hardware has facilitated cost effective WLAN development. Also as speeds of backbone network are increasing speed of WLAN too is increasing. WLAN is an attractive option for small and medium scale industries. It is a technology which will play a major role in field of cost effective and mobile communication in future.

3. ANALYSYS OF PROBLEM:

Wireless LANs have quickly become a significant niche in the LAN market. As adjuncts to traditional wired LANs, they satisfy mobility, relocation, and ad hoc networking requirements and provide a way to cover locations that are difficult to wire. As the name suggests, a wireless LAN uses a wireless transmission medium. Until relatively recently, few organizations used wireless LANs because they cost too much, their data rates were too low, they posed occupational safety problems because of concerns about the health effects of electromagnetic radiation, and the spectrum used required a license. Today, however, these problems have largely diminished, and wireless LAN popularity is skyrocketing.

Communication has been of prime importance to man since ancient time. Various methods have been deployed communication. In early days of voice transfer PSTN networks were used. These consisted of Private Branch Exchange office owned by service providers. These were wired network a copper line connected subscriber home to local office. Local offices were further connected in hierarchical order. Switching was done through hardware like trunk lines processors. Setting up was tedious, time consuming, and costly affairs. Copper wires need to lay down to customer premise. Switching offices need to be setup and hierarchical network backbone need to be established to route calls. All these required lot of time and efforts also this network could not support video traffic.

Next step was towards integration of voice and video traffic. ISDN was employed to achieve this purpose. It could support both voice as well as video traffic. It too was wired technology. They are all digitized, and transmitted at high speed. An ISDN line can carry data at nearly five times the fastest rate achievable using analog modems over PSTN lines. ISDN carry detailed messages back and forth. ISDN offers the means to realize a universal in-box integrating voice, voice mail, e-mail, fax and video images from a single application.

Wired LAN was later employed to transfer voice and video over local area network. It consisted of many configurations like 802.3, 802.4 etc. As it was wired systems connected through it lacked mobility. Also configuring LAN required time. Wires need to be setup to individual PC's. Troubleshooting and maintaining this network was great trouble. Also topology like star could bring whole network down if central hub fails. The wired media in LANs are dominated by a variety of UTP and STP to support a range of local data services from several Mbps up to over a gigabit per second within 100 meters of distance. The early LANs were operating on the so called thick cable to cover up to 500 meters per segment. IN the LAN applications fiber lines mostly serve the backbone to interconnect servers and other high-speed elements of the local networks.

Wireless LAN was employed to remove some of shortcoming of wired LAN. Setting up WLAN was easy and less time consuming. Also systems connected through wireless LAN can be mobile. There is no need to draw costly copper cable to each PC. It is easy to maintain and troubleshoot this system. Popular digital wireless transmission techniques can be divided into three categories according to their applications. The first category is pulse transmission technique used mostly in IR applications. The second category is basic modulation techniques widely used in TDMA cellular, as well as a number of mobile data networks. The third category is spread spectrum systems used in the CDMA, as well as WLANs operating in ISM bands. The main advantage of using wireless LAN is that it provides the ability to change the network infrastructure of an organization easily and without the need for expensive re-routing of cable or the installation of new cable runs.

A WLAN can be configured in two basic ways: Peer to peer (ad-hoc mode) and Client-server (infrastructure networking).

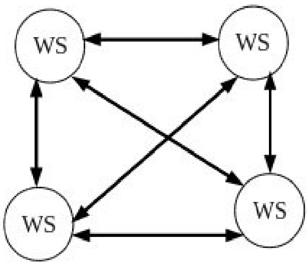

Fig. 1 Ad-hoc networking

The ad-hoc mode consists of two or more PCs equipped with wireless adapter cards, but with no connection to wired network. It can be used to quickly and easily setup a WLAN where no wired infrastructure is available, such as at a conference center or off-site meeting location.

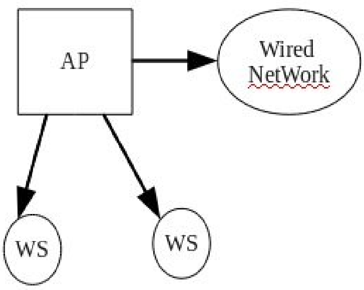

Fig. 2 Infrastructure networking

In infrastrucure networking every WLAN workstation (WS) communicates to any machine through an access point (AP). The machine can be in the same WLAN or connected to the outside world through the AP.

The client-server configuration typically consists of multiple PCs using wireless links to communicate with a central access point that is itself connected by cable to the backbone of the wired network.

3.1 System Block Diagram:

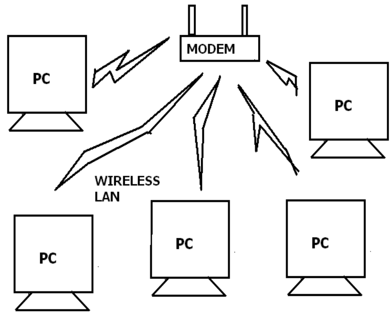

Fig. 3 Wireless LAN used to connect system

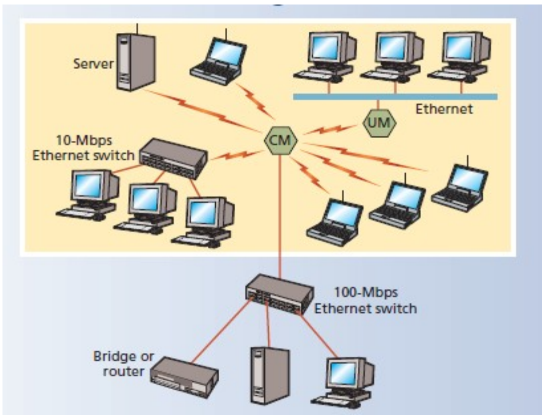

Fig. 4 Single cell wireless LAN configuration

A backbone wired LAN, such as Ethernet, supports servers, workstations, and one or more bridges or routers to link with other networks. A control module (CM) acts as an interface to a wireless LAN. The module includes either bridge or router functionality to link the wireless LAN to the backbone and some sort of access control logic, such as a polling or token-passing scheme, to regulate access from the end systems. Some of the end systems are stand-alone devices, such as a workstation or a server. Hubs or other user modules (UMs) that control several stations off a wired LAN may also be part of the configuration.

3.2 Layering Model of TCP/IP:

a) TCP/IP Protocol Stack:

TCP/IP is the protocol suite upon which all Internet communication is based. Different vendors developed other networking protocols, but even most network operating systems with their such as Netware, support TCP/IP. It has become the defacto standard.

Protocols are sometimes referred to as protocol stacks or protocol suites. A protocol stack term because it indicates the layered approach used to design the networking software.

Layer	Name of Layer	Purpose of Layer
Layer 1	Application	Specifies how a perticular application uses a network
Layer 2	Transport	Specifies how to ensure reliable transport of data
Layer 3	Internet	Specifies Packet format and Routing
Layer 4	Network	Specifies frame organization and transmission
Layer 5	Physical	Specifies the basic network hardware

Table 1 TCP/IP Protocol stack

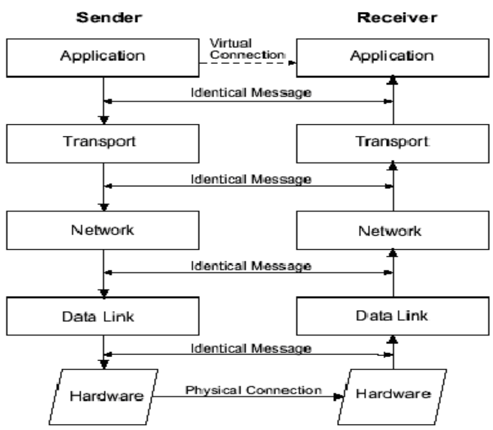

Fig. 5 Flow of data between two computers using TCP/IP

b) Session Initiation Protocol(SIP):

The Session Initiation Protocol (SIP) is a signaling protocol, widely used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP). The protocol can be used for creating, modifying and terminating two-party (unicast) or multiparty (multicast) sessions consisting of one or several media streams. The modification can involve changing addresses or ports, inviting more participants, adding or deleting media streams, etc.. Other feasible application examples include video conferencing, streaming multimedia distribution, instant messaging, presence information and online games.

The SIP protocol is a TCP/IP-based Application Layer protocol. SIP is designed to be independent of the underlying transport layer; it can run on Transmission Control Protocol (TCP), User Datagram Protocol (UDP), or Stream Control Transmission Protocol (SCTP). It is a text-based protocol, incorporating many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP), allowing for direct inspection by administrators.

SIP employs design elements similar to the HTTP request/response transaction model. Each transaction consists of a client request that invokes a particular method or function on the server and at least one response. SIP reuses most of the header fields, encoding rules and status codes of HTTP, providing a readable text-based format.

voice and video stream communications in SIP applications are carried over another application protocol, the Real-time Transport Protocol (RTP). Parameters (port numbers, protocols, codecs) for these media streams are defined and negotiated using the Session Description Protocol (SDP) which is transported in the SIP packet body motivating goal for SIP was to provide a signaling and call setup protocol for IP-based communications that can support a superset of the call processing functions and features present in the public switched telephone network (PSTN). SIP by itself does not define these features; rather, its focus is call-setup and signalling. However, it was designed to enable the construction of functionalities of network elements designated proxy servers and user agents. These are features that permit familiar telephone-like operations: dialing a number, causing a phone to ring, hearing ringback tones or a busy signal. Implementation and terminology are different in the SIP world but to the end-user, the behavior is similar.

c) SIP Network elements:

A SIP user agent (UA) is a logical

network end-point used to create or receive SIP messages and thereby manage a SIP session. A SIP UA can perform the role of a User Agent Client (UAC), which sends SIP requests, and the User Agent Server (UAS), which receives the requests and returns a SIP response. These roles of UAC and UAS only last for the duration of a SIP transaction.

RFC 3261 defines the following server elements:

A “proxy server” is an intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets, and, if necessary, rewrites specific parts of a request message before forwarding it."

"A registrar is a server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles."

“A redirect server is a user agent server that generates 3xx responses to requests it receives, directing the client to contact an alternate set of URIs. The redirect server allows SIP Proxy Servers to direct SIP session invitations to external domains.”

d) *Session Description Protocol (SDP):*

The Session Description Protocol (SDP) is a format for describing streaming media initialization parameters in an ASCII string. SDP is intended for describing multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation. SDP does not deliver media itself but is used for negotiation between end points of media type, format, and all associated properties. The set of properties and parameters are often called a session profile. SDP is designed to be extensible to support new media types and formats.

Thus SDP includes:

- Session name and purpose
- Time(s) the session is active
- The media comprising the session
- Information to receive those media (addresses, ports, formats and so on)

As resources necessary to participate in a session may be limited, some additional information may also be desirable:

- Information about the bandwidth to be used by the conference
- Contact information for the person responsible for the session

In general, SDP must convey sufficient information to be able to join a session (with the possible exception of encryption keys) and to announce the resources to be used to non-participants that may need to know.

SDP includes:

- The type of media (video, audio, etc)
- The transport protocol (RTP/UDP/IP, H.320, etc)
- The format of the media (H.261 video, MPEG video, etc)

e) Real-time Transport Protocol:

Streaming media has certain characteristics as against the normal data. Therefore streaming media warrants a separate transport protocol. Streaming media can be of the following two types:

(a) Real-time delivery of media as in the case of the media-on-demand application. In this case a server may stream a stored media from a file while the client receives and playback the media in real-time.

(b) Delivery of a real-time data. A typical example is a video-conferencing application wherein the captured audio and video are being streamed in real-time.

A very important characteristic of multimedia contents is that they can tolerate small amount of errors. Reason being the human brain can reconstruct the original information from imperfect audio or video signals. However the timing details of the multimedia content should be preserved for the proper playback. Further interactive applications demand a low latency in the transmission of the streaming media.

The default transport layer protocol for the Internet is the Transmission Control Protocol (TCP) [tcp/ip]. TCP is not an appropriate choice for carrying real-time multimedia contents for the following reason. Multimedia contents do not demand the 100% reliability in the transmission that is being offered by the TCP. On the other hand the overhead introduced by TCP (acknowledgment processing and re-transmission) causes large amount of delay in receiving the data and makes the application less interactive. Therefore TCP is seldom used for real-time streaming.

4. APPLICATION:

- It will be useful for small mall scale industries for giving instructions to employees by making audio calls without using internet bandwidth.
- It will be helpful for educational institutions since it is very easy to establish an audio conferencing using which departments can be connected faster and communicated.
- Small organizations which are looking for cost cutting and time saving means for communication can use this.

5. CONCLUSION:

Currently systems are available in market for inexpensive communication. However they have constraints like need for internet connection. In this paper 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.

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