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VoIP over WLAN Networks

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Abstract

It is one of the most technologies in the Telecommunication field .About several research on VoIP the Quality of Service (QoS) still remaining. VoIP simply means to make a call through Ethernet Internet.For eg:-Skype .
On the other hand PSTN is a circuit switched Network based which needed delicated line for Communication.VoIP basic two types of routes
1.Use free VoIP Services
2.Sign up with a VoIP service provider which charge a Monthly free.
VoIP requirment include a broadband modem and an ATA(Analog Telephone Adapter)or IP phone.

Acronyms

VoIP:-Voice over Internet Protocol
TCP:-Transmission Control Protocol
UDP:-User Datagram Protocol
SIP:-Session Initiation Protocol
WLAN:-Wireless Local Network
PSTN:-Public Switched Telephone Network
RTP:-Real-Time Transport Protocol
PKT:-Packet
N/W:-Network

Advantages

- Low expenditure
- Flexibility
- Provide Voice mail and call forwarding
- User can make VoIP calls from one country to Another counter
- Free service to one pc to another pc.
- Easy to install and operate
- Network capacity utilization

Disadvantages:-

- Depends on bandwidth of internet
- IP network does not guarantee QoS for Voice Communication
- User cannot make call during power failed
- Cable fault can make call failed

VoIP system can be configured in the following respective mode:-

- PC to PC.
- Telephony to Telephony
- Telephony to PC

Telephony can be digital or analog type. In the case of analog phone we have to use adaptors which convert the analog signal to digital signal

2. VoIP Internal Components

VoIP contain three components:-

- CODEC:-Coder and Decoder
- Packetizer
- Playout Buffer

When the sender send voice signal and then voice is compressed and encoded into a predetermined format using voice codec. International Telecommunication Union-Telecommunication (ITU-T) developed various standard in voice communication field.

- G.711
- G.729
- G.723.1

In packetization process in which fragment encoded voice into equal size of packet,each PKT contain some protocol header from different layers. Protocol headers add to voice PKT which are knowns as:-
RTP:-Real-time Transport Protocol
RTCP:-Real-time Control Protocol
UDP:-User Datagram Protocol
IP:- Internet Protocol
In addition RTP and RTCP the application layer are designed to Support real-time application,althrough TCP transport layer is used TCP protocol is suitable for less delay sensitive data PKT . This scheme introduced delay as receiver has to notify the sender for each received PKT by sending an ACK on the other side UDP does not apply this scheme ,so it is more suitable for VoIP applications.

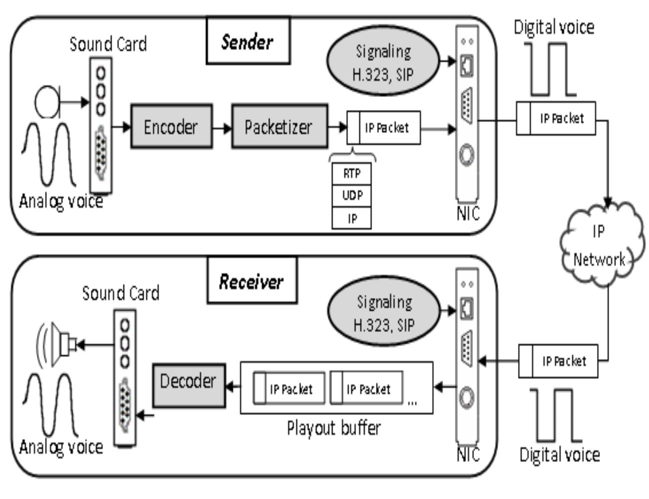

Fig:-1.1 End to End Communication

SIP:-
Session Initiation Protocol and H.323.
The signaling protocols of VoIP. SIP establish end to end close media Streams between the clients.H.323 standardized by ITU-T specially for smoothly working together with SIP and PSTN while SIP was introduced by Internet Engineering Task force (IETF) to support application layer such as telephony.IP addresses can be changed from one session to another, especially in dial-up-case,therefor there is need for a common meeting point shared among user to discover each other at the establishment stage of communication. This meeting point is called as Call Server.

Application Layer	CODEC Voice
	RTP,RTCP,SIP,H.323
Transport Layer	UDP
Network Layer	IP
Data-Link Layer	IEEE 802.11 MAC
Physical Layer	IEEE 802.11 PHY

Fig:-1.2 VoIP over WLAN Protocol Architecture

2. Background of Wireless LAN

The main characteristics of WLAN is simplicity, mobility, scalability ,edibility and cost effectiveness. This technology provides people with a ubiquitous communication in office,hosptials

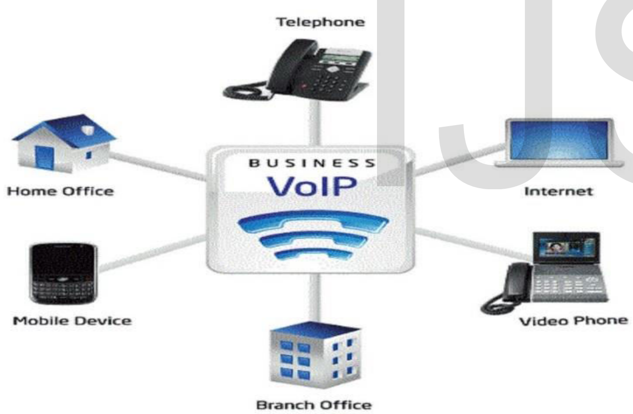

Fig:-1.3 Basic structure of VoIP in organization

colleges, university,factories,airports and stocks market .
It play major function in the next generation wireless voice call networks.it is same as LAN except that the transmission happens via radio frequency (RF) of infrared (IR) and not through physical wires and cables.
Nowadays people demand receiving high speed video,audio,voice and web service even when they are travelling around the office . multimedia application have experienced an explosive development.
WLAN allows to connect and configured in two ways:-

- Infrastructure less mode
- Infrastructure Mode

In infrastructure less mode which is also called Ad-hoc or P2P (peer-to-peer) network, there is no fixed point and each node can directly communicate with all other nodes.
In Infrastructure mode there is the transmission between two or more nodes goes through a third node called Access Point (AP) or Wireless Router.
The terminal communicate with each other through A.P forming a one-hop network. When any terminal wants to send PKT to other terminal PKT would be sent to the AP first which forward them to their destination. IEEE 802.11 WLAN standards protocols deployed.
IEEE 802.11 Protocols operating different frequency:-

- 802.11a support 5GHz and 84Mbps data rate.
- 802.11b support 2.4GHz and 11Mbps data rate.
- 802.11g support 2.4GHz and 54Mbps data rate.

These standard is added by IEEE alphabetic character in each standard .Recently ,the integration of VoIP and IEEE 802.11 WLAN technologies .WLAN cannot provide good service quality for real-time traffic. Therefore deploying VoIP over WLAN poses a challenges in term of performance is expected to be a good as PSTN performance or even better.

2.1 IEEE 802.11 MAC

It is also known as wireless Ethernet it play an important function in future-generation networks. Medium Access control (MAC) sub layer divided into two types:-

- DCF:-Distributed Coordination Function.
- PCF:-Point Coordination Function.

In PCF it uses centralized polling system which require AP as access coordinator.
In DCF it uses carrier Sensing Multiple Access with Collision Avoidance CSMA/CA as the access method.

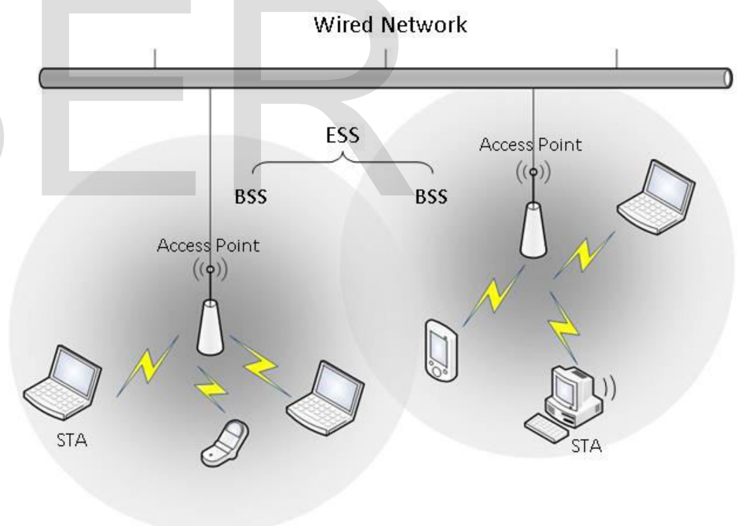

Fig:-1.4 BSS, ESS and Access Point Structure

In CSMA/CA deploys interframe Space (IFS).contention Window (CW) and ACK.IFS is waiting period of transmission over IP based Network. In DCF access method ,STA senses traffic before send or Share PKT over IP-based network.If DCF found medium is sensed busy, sender would be waiting until ready for the transmission this process is known as DCF interfram Space (DIFS).
The sender STA will send Request-to-Send (RTS) to get permission from receiver by sending CTS (Clear-to-send).
CW is number of slots which ranging from 0 to 1..CW stops its Timer while station finds the channel busy due to bursty traffic over Networks.STA can Calculate the random counter interval backoff and choose from CW.A random backoff counter from (0,CWmin).Then CW as :Backoff_Phase=rand(0,CW).slot time,where until busy period When the backoff counter reached 0,the sender starts the transmission. Then receiver sends and ACK to sender.

Access Point:-Wireless Router
ESS:-
BSS:-

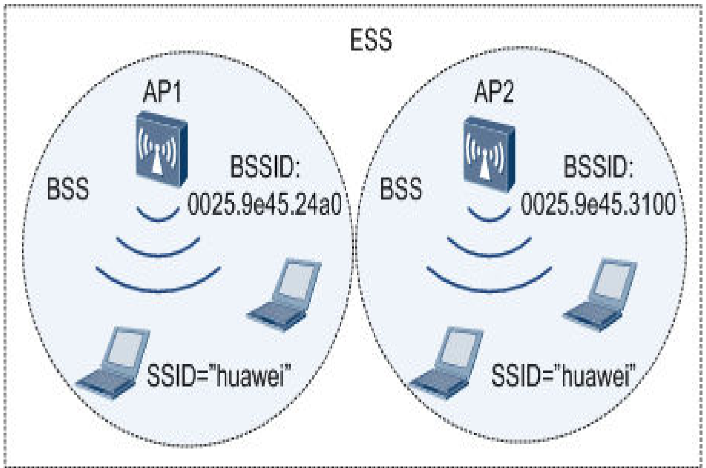

Fig:-1.5 GNS3 Simulator Structure of Huawei

3.Delay

The total time takes since a person, communicating another person, speaks words and hearing them at the other end.QoS should be less Than 150ms for good network connection as defined by ITU G.114 While delay of less than 150ms is defined by the European telecommunication Standard Institute (ETSI).Main reason for delay is Affected by several parameter or algorithm:-

- Delay at the source.
- Delay at the receiver.
- Network Delay

3.1 Delay at the source:-

Codec functions introduce some delays when processing the analog-to-digital conversion.Codec may also perform compression technique for efficiency in bandwidth usage.There is a tradeoff between bandwidth required,and the longer the delay added. Therefore codec add additional delay to the total PKT transmission delay.So there for the delay of the whole process performed at the sender side before transmitting the voice PKT over the network is caused by several component.

Codec Standard	Compression Technique	Bit Rate (kbps)	Encoding Delay(ms)	Loss Tolerence (%)
G.711	PCM	64 (uncpm pressed	0.13(negli gible)	7-10
G.726	ADPCM	24,32 or 40	0.4	5
G.729	ACEIP	8	About 2.5	<2
G.723.1	ACELP	6.4kbps or 5.3	About 67.5	<1

PCM:-Pulse Code Modulation
ADPCM:-Adaptive Differential PCM
ACIP:-Conjugate Structure Algebraic Code-Excited Linear Prediction.
ACELP:-Algebraic Code Excited Linear Prediction

3.2 Delay at the receiver:-

The reverse process that carried out at the sender is performed at the receiver adding more delay; process delay and decoding delay including decompressing delay.Additionally,Playback delay is incurred when playing out the voice stream which includes the jitter buffer delay as well.

3.3 Network Delay:-

Network delay in WLAN is the total delay of both WLAN and backbone network. Queuing, transmission and propagation are other component of network delay. Router node causing queue delay. The delay in the physical media of the network, while transmission delay includes router's delay and MAC retransmission delay.

4. Jitter

IP n/w does not guarantee of PKT delivery time which introduced variation in transmission delay this is known as Jitter. It has negative effect on Voice quality. The PKT of same flow are not received at the same time the jitter buffer are introduced to diminish the jitter effect and make the conversation smoothly as it holds a group of PKT in queue before playout.the buffer queue size can be fixed or adaptive which varies based on network condition, voice character for better performance. Buffer jitter adaptive techniques perform better as it reduces the possibility of buffer overflow and underflow.

In overflow the number of PKT recived is getting larger than the buffer size as a result buffer discard PKT that cannot hold.
In underflow buffer occurs when some PKT are needed for playout but buffer is empty. Several adaptive buffer algorithms are introduced to adjust buffer size in order to improve quality of VoIP.

4.1 ECHO

Echo is the term of the reflections of the sent voice signals by the far end. Echo could be electrical echo which exits in PSTN N/W echo Which is an issue in VoIP N/W.This signal is generated due to the conversion between the 2 wire lines and other circuit of other type in a hybrid environment. The echo is affected by delay and it becomes noticeable when it is loud and delayed mostly when the round trip delay is more than 50ms.It is exceed up to 100ms.
Echo cancellers are used to cancel the ECHO.

4.2 Throughput

The transfer bit rate of IEEE 802.11b 1Mbps,2Mbps,5.5Mbps and 11Mbps.the standard dynamic rate switching to improve performance of wireless link.
The throughput is achieved between the range 50% to 70% of the transmission rate which is low comparing to Ethernet throughput Which archives 80% to 90% of the transmission rate
Two method are introduced for improving the throughput :-

- Aggregation Technique.
- Adaptive technique.

5. Capacity of VoIP over WLAN

No	PKT interval(ms)	G.711	G.729	G.723
1	10	6	6	
2	20	11	13	
3	30	15	19	19
4	40	19	25	
5	50	22	31	
6	60	25	37	37

The maximum number of VoIP connection over 802.11b

6. Security to VoIP

VoIP has many security issues but PSTN has high level of security. But some similar issues are found in VoIP and PSTN. Eavesdropping and toll fraud. In Eavesdropping which listening and changing conversation is possible for attacker.it change the conversation play back. For example:-"YES" to "NO or take advantage by asking username and password.In Toll Fraud the attacker is making free calls especially for high cost long distance calls, fraudulent calls are billed to a victim and it possible due to miss configurations.Some threats and risks inherited from this loosely network to VoIP.Dos Denial of Service the attack which tries to make resource unavailable or impede a service from functioning well. VoIP DoS attack aimed to bring a service down. VoIP runs voice over stream of IP PKT.SPIT (Spam Over Internet Telephony) which trouble only VoIP like VoIP spam is related to voice-mail inboxes. The spammers will send many unwanted message and audio-commercials to make voice-mail inboxes full to need extra voice storage capacity. Protocol Fuzzing (malformed messages) is other issue belongs to IP applications. It generate random or sequence input data to inject them in an application normally fuzzing is used for debugging and testing IP application. Protocol fuzzing target is a protocol like SIP stack protocol. Attacker uses this method to finds vulnerabilities and creates various PKT type that aimed target system to process these PKT instead of normal PKT. This may cause system failure or crash, buffer overflow application delays and infinite loop.

User uses secure VoIP, last security issue WEP (Wired Equivalent Privacy) the weakest attempt, WPA(Wi-Fi Protected Access) is stronger and WPA2 is enhanced WPA.IEEE 802.11 is suggested to have secure WLAN. Firewall, NAT and IPsec is another well Known security mechanism.

7.Conclusion

This paper has a survey on VoIP over WLAN on single Ethernet connection. We use single Ethernet for data and voice communication.Adavantage and disadvantage of VoIP.Jitter play a main role on voice quality therefor jitter buffer for smooth play out of PKT.Echo and throughput.Capacity of VoIP capacity over WLAN and number of calls for different voice and codes .

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