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Enhancement of Voice Technology over Ad-Hoc Network (MANET)

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ABSTRACT: Voice over IP is most prominent developing application layer protocol. Real time voice communication is a basic necessity in specially appointed ad hoc network. While ad-hoc network (MANET) effectively appropriate in little scale situation. In this paper, we distinguish a few factors that impacts voice quality in MANET for voice correspondence AODV and SIP protocol is utilized. In this research, to maximize the execution of MANET during voice transmission. We choose a few methodological techniques and parameter, for example, audio codec, velocity, method of media access, buffer size etc in optimum an efficient way. This paper, a cross – layer approach is intended to enhance the call setup execution of SIP-based VOIP over AODV – based MANET. Cross layer approach is utilize the SIP execution metric to enhance the call setup time in term of packet loss, network load, delay time by modify Time-to-Live parameter. For simulation, we have taken OPNET14.5 addition simulator. This paper purposed another quality of services VOIP calls over in MANET and to maintain the highest number of calls with safety and security. Utilizing this approach, the SIP singling execution expanded by 30-40% when contrasted with default appreciates ADOV MANET execution.

KEYWORDS: MANET, AODV, SIP, VOIP, OPNET, CLAODV, TTL

I. INTRODUCTION

MANET is an independent system which is gathering of self composed and self – arranged that gives adaptable techniques to communication between portable base less devices with no prerequisite for settled surroundings no organize services support. For example, Mobile devices, workstations and PDAs. The nonappearance of settled foundation implies that the hubs specifically associated or discuss straightforwardly with one node to another node arrange in a peer-to-peer system. There are unmistakable confinement techniques for wireless network. For instance high error rate, power restrictions, bandwidth constraints etc. and so forth anyway these couldn't imperatives the increase of wireless technology. Versatile specially appointed system MANET is a one of a kind motivation behind community for industry and researchers from all around the world. This techniques has went with its own flavors it is definitely not hard to pass on in catastrophe zones and for crisis tasks.[2]

MANETs may be produced by a sort of wireless devices with radio transmitted and beneficiary. For instance nodes keep up great radio interfaces. Since these nodes are remote devices and might use battery control, they are absolutely autonomous. In this MANET, they may move wherever, While up 'til now passing on each other. The two nodes adaptability and heterogeneity are ordinarily qualities of MANETs. In this sort of system, route packet that have a place with various transmissions. Voice over IP (VOIP) is an advancement for the movement of voice communication and multimedia session over an IP (internet protocol) network, for example, the internet. [5].

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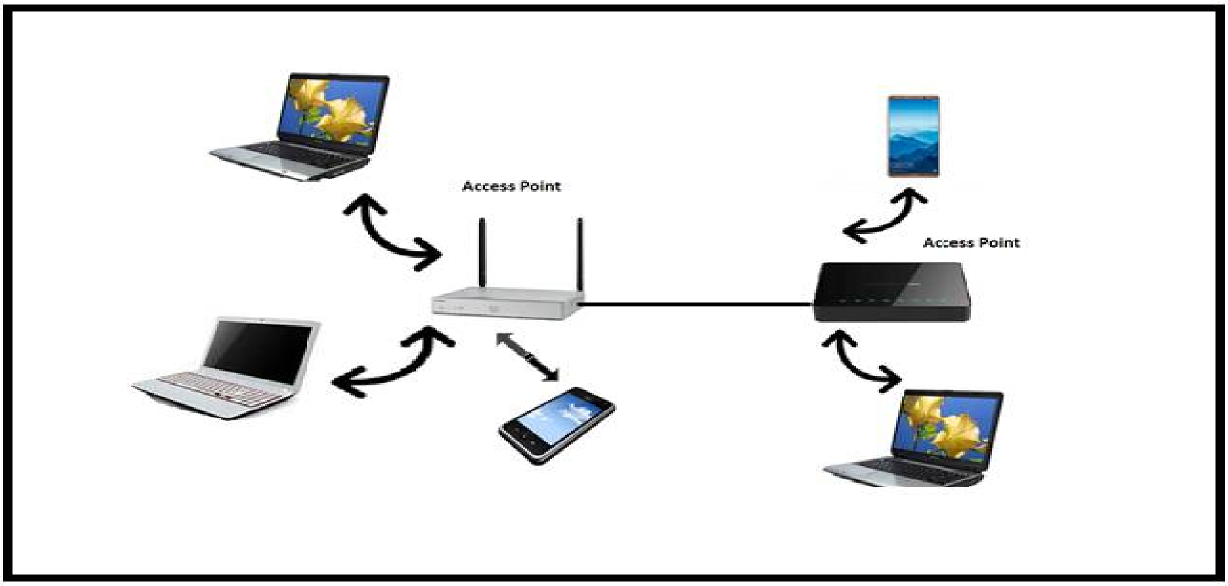

Fig 1 Infrastructure-less (Mobile Ad-hoc) Network

The session initiation protocol (SIP) today is seen as the standard protocol for intelligent media signaling, and the result is an amazingly non particular protocol.SIP is indicated by the IETF in RFC 3261. From essential perspective, SIP is content – based protocol the same as HTTP use as a piece of web service.SIP can exchange differing sorts of payload with different encoding. SIP is a stateful convention that support both UDP and TCP as transports. From the framework perspective, the dynamic undertaking of SIP and most other signaling protocols is in a general sense the same as FTP, yet without the benefit of "latent mode". This suggests SIP organizes dynamic UDP port sets in the two completions for the RTP media streams. Other than signaling, SIP is similarly used for texting.SIP specific is likely the longest detail anytime released by the IETF and is extended further in various different determinations. As opposed to various protocol IP protocols, [5]

II. RELATED WORK

Yi deng et.al. [3] in this paper, author contention on underground mines and QOS assessment of various voice codes over specially appointed systems. Here, author looked at G.711, G.723 AND G.729 three diverse codec on premise of execution. by utilizing parameters, for example, bit rate, framing, intervals, and payload packets. In their work, author has taken ns2 simulator along parameters Mac layer, organize layer voice codec, VOIP term, simulation time. The experimental outcomes got from execution of various sound, codec's in underground mines. Author derived that with the size of situation and VOIP session QOS on VOIP decrease G.7231 is best codec as far as giving an acceptable MOS score. In interim, G.711 isn't appropriate for VOIP over ad hoc network in underground.

Weheb et.al [6] in this paper, author assessed QOS and the present MANET routing protocol. For VOIP, HTTP and FTP application utilizing distinctive simulation models. Here, author enhanced the execution by utilizing number of parameter network identity, load traffic and node mobility .in this author has taken Exata 3.1 test system which is utilized to assess the execution in various situation. Here, authors comprise of various measurements for various movement write, throughtput, end to end delay, jitter and so on. After simulation author reason that MP-OLSR best execution over DYMO and OLSRv2 as far as high thickness, high versatility and high congestion.

Adil Mazhar Qureshi et.al [7]: here author, compressively simulation based examination of routing protocol for MANET. Here, author investigation execution with various Mobility Models to acknowledge reasonable situations. In this paper author's e have assessed the effect of differing rate of nodes by general throughput of the system. here ,author's

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has taken ns-3 test system with number of parameter number of source, number of sinks, total number of nodes, misfortune demonstrate etc.Here simulation has improved the situation a wide speed go from 5m/s – 160 m/s. in this author separated the speed runs in 3 sub classes like. low, medium and high. the execution at low speed of OLSR is better; yet as the speed of the node is expanded AODV protocol begin better QoS. For Random Walk OLSR protocol has high PDR anyway by expanding speed AODV gives better outcomes nearly. After the simulation author's reasoned that by and large AODV perform superior to the next OLSR and DSDV calculation in variation portable condition under various speeds.

Anupama Sharma et.al[8] in this paper, author investigation the multicast routing protocol in light of QOS. In this present, author's investigation the execution is done alongside their component and focal points. Here, author Performance examination in light of their network architecture, QoS parameters, qualities .by this paper new analysts effortlessly audit about directing data which is introduced in unthinkable shape . in this work, creator's investigation some ongoing QoS based multicast way discovering calculations for remote specially appointed systems . after investigation ,authors presume that , a few regions where analysts can center in their further research. in this paper will be utilized by new scientists in creating routing protocols for MANETs.

III. VOIP

Voice over Internet Protocol (VoIP) is the dominant open protocol for Internet telephony. The components of a standard VoIP exchange.[9] When Raj and Deepak wish to communicate by a phone call, their VoIP telephones set up a sound connection utilizing the Session Initiation Protocol (SIP) by means of what is named the signaling way. As VoIP endpoints are regularly versatile or situated on unique IP addresses, signaling way exchanges are mediated by proxies – specialist service provider or corporate VoIP signaling gateways that get and forward messages for the benefit of their customer endpoints. When Deepak raj wants to communicate to each firstly their contact her SIP proxy who forwards the call invitation to Deepak's SIP proxy, as only Deepak's proxy knows the current IP address of Deepak's endpoint. The body of the welcome contains both information about Raj and the RTP deliver where she hopes to get sound (or other media streams) from Deepak, should he acknowledge the call. Deepak's acknowledge of the welcome contains the RTP address of where he hopes to get sound from Raj which permits an immediate, bi-directional media way between their endpoints. Raj simply wants to conversation to Deepak but the network requires that the communication be addressed to the IP address of Deepak's phone. The entire infrastructure in the Figure 2, together with the several services, devices and DNS name enrollments that are not appeared, exist exclusively to delineate the client/application's reality view into the system's reality see. One in number driver for content-oriented networking is that this translation (commonly alluded to as middleware) isn't required. Information ought to rather flow directly from producer to interested consumer. Our VoCCN model accomplishes this.[10]

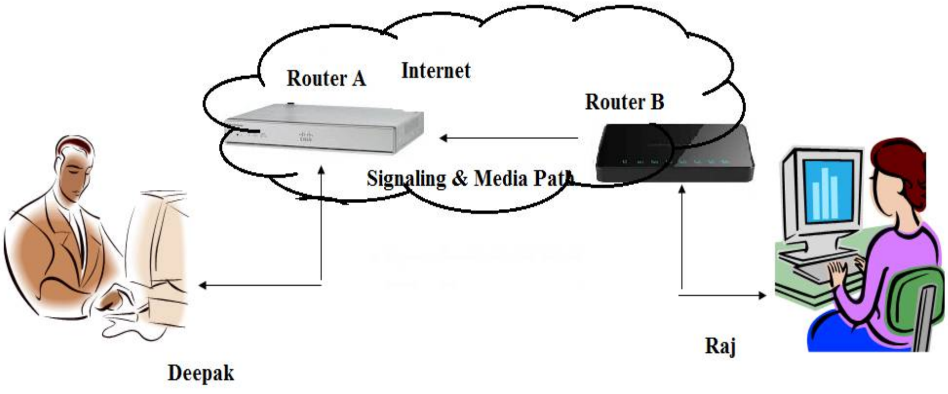

Fig2 Voice-over-CCN data flows

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IV. PERPOSED METHODOLOGY

Cross layer approach

Cross Layer approaches is use to refresh the controlling coordinating routing to upgrade its application execution over MANET. Cross layer is utilized three way handshake approaches. Here, three way handshake structures are related for call setup. Cross layer approach is related with refresh the coordinating execution by moving message in cross layer with the structure layer adjust the AODV parameters by utilizing it isolated information in the application layer. Performance of call setup performance depends on the SIP initiation messages sent by the sender node and received by the receiver node. For MANET systems with variable node capacity like moderate to high node capacity, variable hop numbers and node mobility, performance may increase by setting the routing parameters to the required level using the SIP performance matrices. In this approaches SRD calculation is done for all calls those initiated by the caller node UA(A). All the call initiation signaling pass have to pass through a server known as SIP server. The basic function of SIP server to recognize all messages and parameters for INVITE messages along with the caller to determine the SIP signaling performance, as shown in Fig. 3. The call setup time calculated as the time difference between TIn1 and TA3, where TA3 is the time of receiving the call acceptance acknowledgment 200 OK by the callee UA(B). Under this implementation, SRD calculation is the time difference between the INVITE message, sent at time TIn1, and Tx3, the time at which the callee's response message was received by the UA(A) for the call invitation, as shown in Formula (1). Under this approach, another system used to call setup is three-way handshake system which define the basic SIP signaling flow between the call entitie/nodes, and based on result the SRD SIP performance metric can applied. The proposed method is applied to the SIP server side and the caller node side as well. [1]

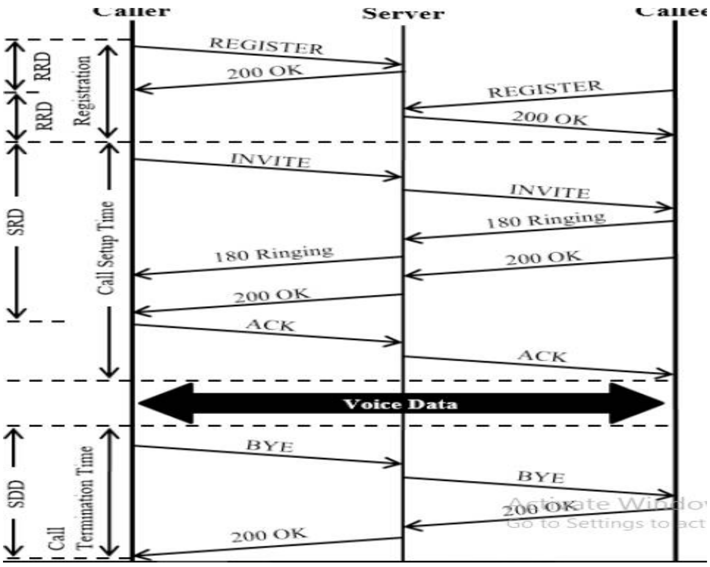

Fig 3 call setup signaling flow for the proposed CLAODV[1]

V. SIMULATION AND RESULTS

Simulation is completed in OPNET test system. Here 110 portable nodes are conveyed in a wireless environment to upgrade execution of packet loss, Network load, and end to end delay time . The two situations were kept running for a day and age of 26 minutes, which extended from 0 to 1600 seconds as appeared in the outcome diagrams. . From that point onward, we investigate and look at inside every situation and furthermore the two situations in light of their call setup time, Network load, Packet drop. Fundamental parameters utilized for experimentation with OPNET test system. Region for correspondence is 100km x 100 km with 110 versatile nodes. The execution examination of two situations in

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term of Packet Drop, Network Load, and Call Setup Delay time is clarified in figures. Here, table 1 demonstrates the reproduction parameters taken during simulation.

TABLE 1. SIMULATION PARAMETERS IN OPNET

Parameter	Value
Examined Protocols Cases	VOIP with SIP
Number of Nodes	110
Types of Nodes	Mobile
Simulation Area	100*100 km
Simulation Time	1600 seconds
Mobility	Uniform(10-100) m/s
Pause Time	15 seconds
Performance Parameters	Packet loss, Delay, Net.load
Trajectory	VECTOR
Long Retry Limit	3
Max Receive Lifetime	0.5 seconds
Buffer Size(bits)	25800
Mobility model used	Random waypoint
Data Type	Constant Bit Rate (CBR)
Packet Size	512 bytes
Traffic type	FTP, Http
Active Route Timeout	5 sec.
Hello interval(sec)	1,2
Hello Loss	6
Timeout Buffer	2
Physical Characteristics	IEEE 802.11g (OFDM)
Data Rates(bps)	12 Mbps
Transmit Power	0.005
RTS Threshold	1024
Packet-Reception Threshold	-95

A. Packet Loss

Packet loss is characterized as the percent of packet dropped at the recipient limit preceding information stream playback. Despite the fact that VOIP application endures packet loss up to 10% or a parcel loss of 1% still influences the nature of the VOIP calls. VOIP is continuous real time audio services which utilize the untrustworthy UDP protocol. Subsequently, the recovery of any lost packet inconceivable. In figure the chart speaks to the packet misfortune in bits every seconds. The x-pivot signifies the simulation time in minutes and the y-node indicates packet loss in bits every seconds. It can be obviously observed, that the whole system packet loss is diminish once the proposed brought together instrument is executed. Likewise, the condition of the packet loss has diminish more than the typical VOIP subsequent to executing the calculation. packet loss esteems at each second appeared in figure.

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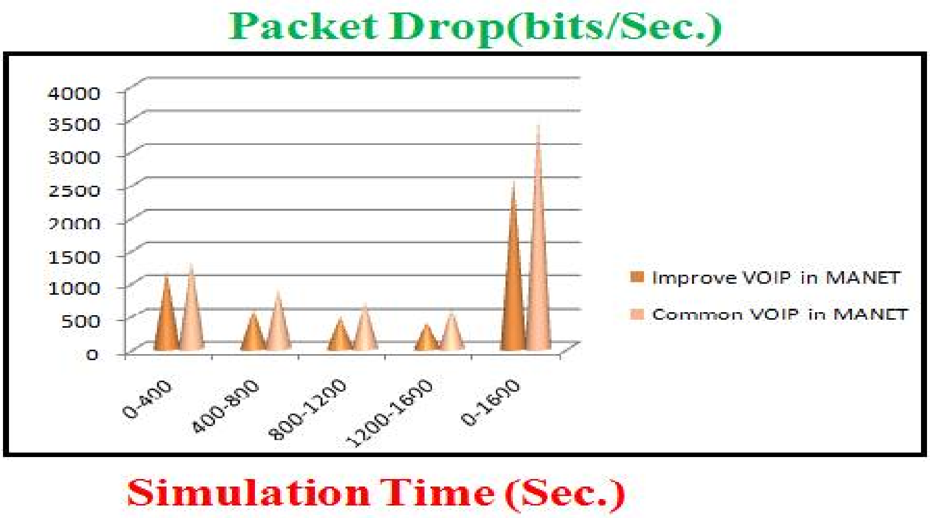

Fig 7. Packet loss results

B. End –To-End Delay

The packet end to end defer is the normal time that packet take to cross in the system. This is the time from the generation of the packet by the sender node up to their gathering at the goal and is communicated in a moment or two.

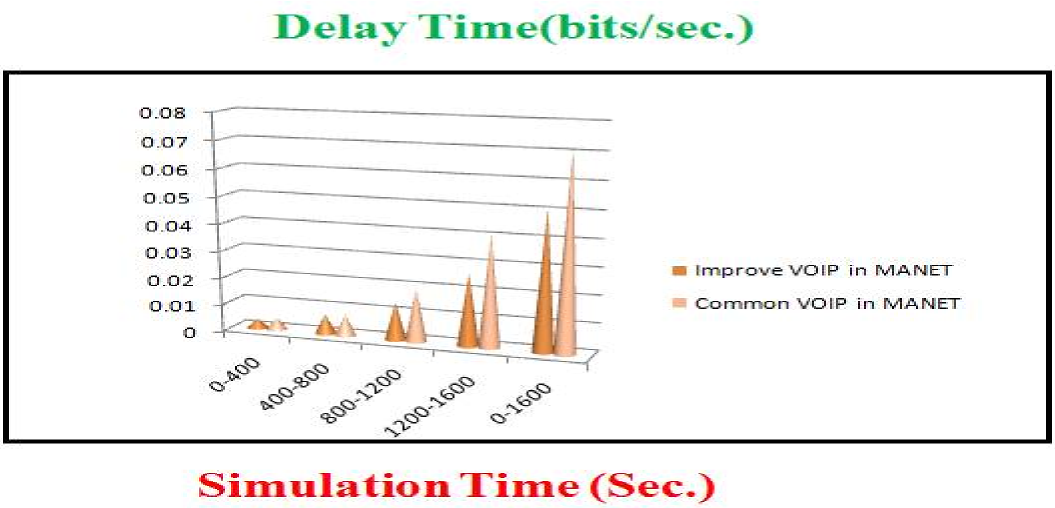

Fig 8. End-To-End Delay results

Thus all the delays in the system are called packet end-to-end delay. It incorporates all the delay in the system such propagation delay (PD), processing delay (PD), transmission delay (TD), queuing delay (QD).

In figure the chart speaks to the conclusion to end defer time in bits every seconds. The x-pivot indicates the simulation time in short order and the y-node means end to end delay time in bits every seconds. Here, we looked at two situation, first situation which is Explain regular VOIP in MANET and Second situation which is Explain enhance VOIP in MANET.

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C. Network Load

Nnetwork load is measure of information moving over a system at a given purpose of time. Network is for the most part epitomized in network packet, which give the load in the system. System stack is the primary segment for network load estimation, network load control and simulation.

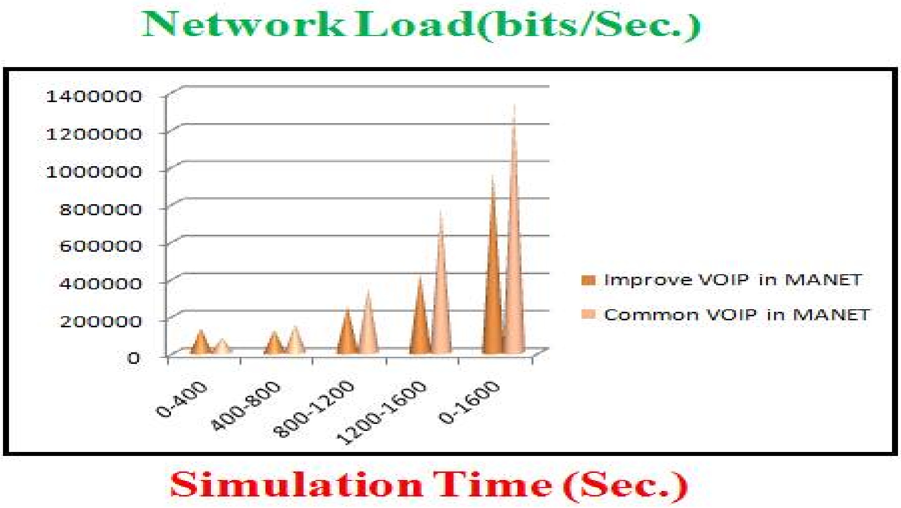

Fig 9. Network Load results

V. CONCLUSION AND FUTURE WORK

In this, examination was exhibited in regards to the call setup execution of SIP based VOIP over the MANET AODV. To upgrade the execution of SIP signaling, another outline called as cross layer AODV MANET configuration was proposed. What's more the improved AODV directing parameter was utilized that utilized longer TTL esteem for RREQ messages, that is help to upgrade the call setup execution. Utilizing this approach, the SIP singling execution expanded by 30-40% when contrasted with ADOV MANET execution. We likewise execute OPNET test system to upgrade the execution regarding packet loss, delay time and network loads figured in light of setup of 110 nodes. The packet loss execution expanded by around 30-35%, network load and end to end delay execution expanded by 25-30%.For future, new approach will configuration called as CLAODV for upgrade the call setup execution. This approach will dissect CPU effectiveness and transmission capacity use too. Likewise, the matchup between call setup execution and CPU execution will considered while setup.

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