ISSN(Online): 2320-9801
ISSN (Print) : 2320-9798

International Journal of Innovative Research in Computer and Communication Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijircce.com

Vol. 6, Issue 3, March 2018

Review on VOIP over MANET by Utilizing SIP

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ABSTRACT: In this work, voice over IP (VOIP) PROTOCOL transmission in mobile ad hoc network (MANET) reviewed .All though a lot of research and implementation have been performed for voice transmission over ad hoc network. But very little is done for SIP an AODV in ad hoc environment .in this paper, we analysis several aspect of VOIP. In this paper, also explain various advantage and limitation of AODV routing protocol over ad hoc network using VOIP technology. At last in this paper, we review the existing research on SIP protocol over MANET using VOIP technology and AODV protocol in mobile ad hoc network through VOIP.

KEYWORDS: MANET, VOIP, SIP, NS2, CODECS, AODV

I. INTRODUCTION

a mobile ad hoc network (MANET) is a network of mobile devices associated remotely . every one of devices can work autonomously and move to any of HEADING, ALLOWED to interface with any devices. every node in MANET functions as switch for alternate nodes. as nodes are self –sorted out require higher security over network that is the reason security is torment region for specially appointed network. in MANET, nodes communicated by using multihop route. here, multi-hop routes change as indicate by topology which is resolved utilizing protocol. for example DSDV, DSR,AODV and ZRP. From all these protocol AODV protocol is better from DSR and DSDV in end to end delay protocol and its very efficiently protocol in case of limited bandwidth.

on the other hand voice over internet protocol (VOIP) is a type of transmission that enables any individual to make telephone brings over a broadband internet connection. VOIP get to more often than not enable the client to call other people who are likewise accepting bring over the internet. VOIP can also provide a unique solution that enables the exchange of voice signal over internet connection routing the traditional telephone like. one main positive point of VOIP that client can longer distance calls at very cheap prices which include calls to other nations in distancing parts of worlds.

II. RELATED WORK

Carlos soria-lopez et.al. [1] in this, author petition in MANET conventional VOIP play out delay control algorithm. Today's on internet, some factor crash on voice quality of VOIP session is jitter, delay and packet loss. the dispatch of VOIP period recreation a very important role. by used a routing protocol in MANET and other hand play out delay control algorithm recreation another mandatory role on VOIP and helpful in reduction jitter effect and increase the VOIP session interactivity. In this paper, author has taken ns2 simulator. For simulation purpose author has taken various simulation parameter such as packet size, rate and burst time, idle time and phone call duration. in this paper, author has taken with AODV and DSDV protocols. Here, author used two different algorithms. after the result, author derived that DSDV perform much better than AODV with regarding of the packet delivery ratio.

Raman sanchez-iborra et.at. [2] In this author argument on VOIP services by accomplishment appraisal of batman routing protocol on QOE viewpoint. MANETIS represented as decentralized topology and ebullient maturing at low

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cost distribution. MANET nature network has a dynamic for improving performance and service employ a appropriate routing protocol. BATMAN protocol which proposes of avoiding the interchanged routing information number of node is one routing protocol that has been a devoted notable research from last few years. in this paper, author focuses on appraisal of batman protocol on VOIP service at low power consumption and QOE perspective. Here, authors enhance the accomplishment of batman protocol in physical layer by using fading and number of density node and node mobility. after the result, author derived that performance of batman protocol are compared with widely used OLSR routing protocol .after the result author derived that energies neither saving node batman nor OLSR is current implementation are suitable for VOIP traffic.

ParamjitSingh et.al. [3] in this paper , author's analysis neuro fuzzy inference system over MANET to predict the optimum routing under stressful condition using VOIP technology. In their work for evaluation purpose author had taken 80 nodes1250 into 1550 meter sides square simulation area with AODV protocol. after the simulation result AODV provides higher throughput in small area but in the stressful situation the performance of AODV decreases as mobility increases.

tahsin Arafat reza et.al.[4] in this paper, author's purposed quality aware adaptive security scheme they use h.264 video coding will 30millsecond that counters the effect of delay overhead by cryptography and multimedia services. for performance evaluation, authors had taken ns2 simulation tools. in their work, crypto threshold. an algorithm to determine the value of crypto threshold is purposed.

Emerson p et.al. [5] in this paper, author focused on real time voice communication in MANET . here, author recommend the correct, GPS does not depend on free situation calculation of framework. The basic principal of committal of this work is to characterize and register relative position of node in crucial system unescorted by employ GPS. in this, authors classify how the proposed methodology can be associate to vast amount particularly selected system. here, author proposed and proposed algorithm which use GPS free positioning it does not depend on GPS devices and it show the position information without infrastructure and without GPS device. here, author main contribution in defining position of node in an ad hoc network without using GPS for real time voice communication.

Sonal Telang Chandelet.al. [6] in this paper, author in wireless lan experiment analysis of various protocol on VOIP traffic with different codecs. Variants protocols are evaluated for considering the impact of node density. the enhance the number of node diminish voice quality. here, different QOS parameter such voice QOS parameter such voice QOS parameter such voice MOS, the end to end delay, voice jitter, throughput and packet delivery ratio are taken to improved the execution. in this paper, author has QOS parameter such voice MOS, the end to end delay, voice jitter, throughput and packet delivery ratio are taken to improve the execution. in this paper, author has taken cyber2.1 simulator and emulator employingcyber2.1 simulator and emulator employing parameter. simulation run time studied protocol, area, network type, data rate, no of nodes, node position, node density, queue type. after simulation routing protocols DSR and AODV with g.729a codec both 50 node and 100 nodes show the best result. Authors conclude that OLSR is better on small sized network. otherwise DSR are well performed as compared to AODV for large scale network. after that author derived that three codec g.729a codec is better than other different codec with respect end to end and c end to end and voice jitter.

Md.Golam Kaosar et.al. [7] in this paper, author analysis energy aware routing protocol through several audio codecs (G.711,G.729 AND G.7231) after simulation result author conclude that G.729 codec is more suitable for voice transmission over MANET with respect of latency.

Rekha G et.al.[8] in this paper, author main purpose is to simulate a distributed network model for voice communication in search and rescue operations .sensor nodes are deployed to relay facts approximately the environment. here, author has taken Zigbee employ network simulation time, channel propagation model, initial energy of nodes , ad hoc routing protocol ,MAC layer, number of nodes, traffic type, traffic rate. To enhance the performance of the network, different parameter such as throughput, average delay and packet loss ratio for different voice data rates. After that result, author derived that DSR perform good for static or low mobility network. For this search and rescue operation can be simulated for low mobility network.

MuhammadAamir et.al.[9] in this paper, author present a new protocol VQAPM to improve the voice quality in VOIP eventual when conveying nodes are the piece of some MANET. here, authors utilize two particular parameter of voice communication over IP. In is voice codec and other is ip packet length. in this paper, author has taken a node in MANET is chose as MVCH (MANET voice communication head) that applies the proposed algorithm to decide on perfect combination of bit rate and packet length. in their work, author has taken ns2 simulatior.for simulation purpose

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author has taken number of parameter such as packet delivery ratio. After simulation result, authors conclude that VQAPM increase voice quality in a number of parameters.

ChristianGottron et.al. [10] In this paper, authors present a new protocol VQAPM to improve the voice quality in VOIP eventual when conveying nodes are the pieces of some MANET. Here, authors utilize two particular parameter of voice codec and other is ip packet length. in this paper, author has taken a node in MANET is chosen as MVCH (MANET voice communication head) that applies the proposed algorithm to decide on perfect combination of bit rate and packet length. in their work, author has taken ns2 simulator. for simulation purpose author has taken number of parameter such as packet delivery ratio, end to end delay, jitter and queue delay. After simulation result, authors conclude that VQAPM increase voice quality in a number of parameter.

Muhammad shaffatual Islam et.al.[11] in this paper, author improve the performance in mobile ad hoc network(MANET) by various vocoders. Mobile ad hoc network is best suitable option for convention centre. the attainment of VOIP services depends on different voice codec G.728,G.711.G.723,G.729,G.726.GSM-EFR,GSM-HR. the execution of seven vocoders G.728,G.711.G.723,G.729,G.726.GSM-EFR,GSM-HR has been compared and analysed by two different medium and large scale scenario. HERE, author evaluated the performance with number of parameter such as average, end to end delay, jitter MOS and average delay variation. we found two scenarios under load condition that g.711 performing better in medium scenario of 25 node and GSM-EFR well performed in large scenario. In this paper, author has taken OPNET simulator for various vocoders. after the result, author derived that high data rate codec have excellent performance.

III.**MANET**

MANET is a self arrangement network having wireless mobile node communicating with other nodes by utilizing multi- hop wireless links without any centralized control.MANET architecture consist mainly four networks bluetooth,ad- hoc sensor networks smart ad-hoc communication networks wireless fidelity networks which networks which effect the communication devices[4]

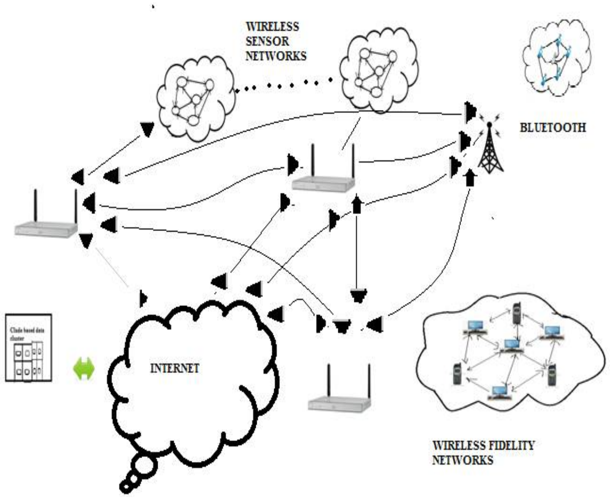

Fig 3.MANET Architecture

Ad hoc sensor network: - ad hoc sensor network consists of regular nodes and super nodes. they have square with chance to access to the system, yet have diverse sending and forwarding .the sending data onto the super nodes for that regular nodes is responsible for monitoring the surrounding environment. Gateway is a way by which super nodes connect other type network and forward data onto the destination ad hoc sensor networks power control has been done by the unnecessary wireless communication and super nodes choice mechanisms to acquire effective information sending network topology. For example, heuristic algorithm DPSO, GRAS mechanism, WCMprotocol based on cognitive learning, larp protocol in underwater sensor networks.[15]

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Smart ad hoc communication network-smart ad hoc communication networks consist Bluetooth, infrared and other personal network. The temporary communication network is universal with the increasing of intelligent terminal devices. Bluetooth is basic and advantageous short-run communication protocol that broadly connected in smart phones, game devices, and smart home and so on. Bluetooth is a radio innovation for wireless individual zone organizing working in the 2.4 GHz ism frequency bond and enables smart terminal devices to associate. Cellular network can be sent to the message from Bluetooth to cloud based data canter whenever and wherever.[15]
Wireless fidelity networks:**-** wireless fidelity networks are very imperative for individual communication and local personal network, which refer to any type of wireless network. In wireless fidelity networks using wireless communication innovation, nodes can be linked to each other without cable. The smart terminal device interface internet by utilizing wireless fidelity routers. the network can help a different variety of standard network communication protocols which have high devotion and transmission speed. For example:- GSAA,CARP professional protocol.

IV.**VOIP**

Voice over internet protocol is an innovation that permits passing voice signal through packet switched lines as opposed to a simple telephone line or analog phone line.[1] It is notable that VOIP applications have inflexible QOS necessities for a steady and reliable communication. The QOS on VOIP is particularly delicate to end to end and packet loss rate. As well as, the kind of voice codec has the effect on the QOS on VOIP because it's initial function of performing analog/digital voice transferred and digital compact.[14][2] In underground mines, Ad hoc networks are special class of mobile ad hoc network provision real**-** time information. As of late, the voice over internet protocol system depend upon on session initiation protocol (SIP) has grown quickly.

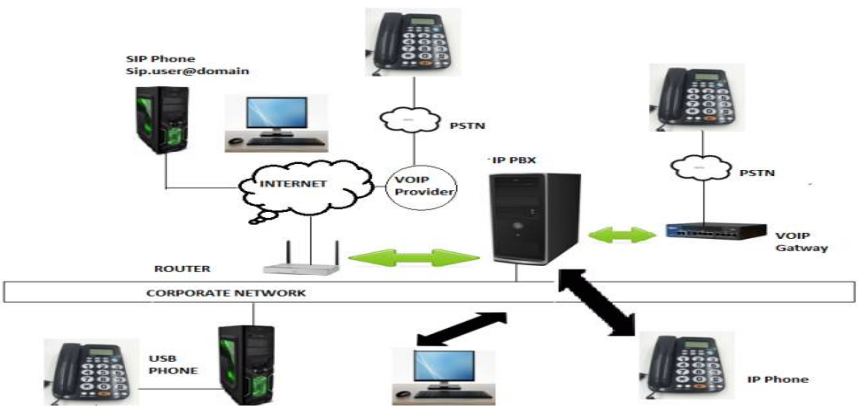

Fig4. VOIP Architecture

Endless supply of a simple voice on the phone, voice gateway initially digitized signal and compact the new digital signal in the form of compact standards information pieces know as IP packets. [1]They are sent over the internet to the passage of the voice gateway, where the procedure is turned around with this innovation .it is conceivable to make three unique varieties of calls: PC to PC, PC to telephone and telephone to telephone. The procedure of internet communication begins calling –the system computer, microphone and headphones, needing to associate with the called party**-** The telephone from public telephone network. A Source of internet calling taking the required software associated with the called party and a phone number to the ISP that supplies services (VOIP). Utilizing a microphone, called party at that point converse with him and voice signal is exchanged with the voice gateway, where it is

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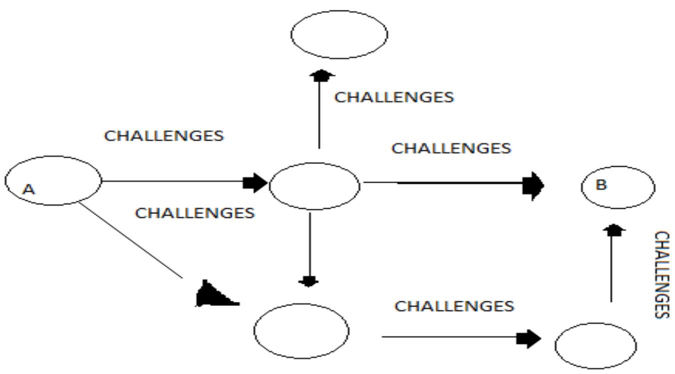
digitized.[16] From here, IP packets are exchanging with the internet out and about, which is controlled by the voice gateway to the supplier until it points when it achieves the remote voice gateway and then exchanges the voice to the neighbourhood PSTN to the called party[12]. In VOIP main issues that still remaining a challenge that how to address and reach the user on a computer located anywhere in the world.

Type of VOIP protocol: -sip (session initiation protocol):- is an interchange protocol for signalling and controlling media communication session in uses of internet communication for voice and video calls, in private ip phone telephone systems, and in addition in messaging over internet protocol (IP) systems.sip is a text based protocol, combine many elements of hypertext transfer protocol and the simple mail transfer protocol.[16]
h.323:- one of the principal VOIP call signalling and control; conventions that discovered far widespread implementation .since the advancement of newer, less complex protocols. For example, MGCP and sip, h.323 organizations are progressively constrained to conveying existing long haul network traffic[16]

V. SIP FOR MANET

In the SIP INFO packets inserted HELLO packets as header. When a node does not know the best path to reach a destination. In the SIP INFO request packet consist two new headers: Challenge and challenge response, challenge packet just contains the destination node that must be come to and desired connection parameters.[13] Rather, a challenge response contains a few data onto the way that should be considered and the association parameters that can be ensured. So that sources do not contain information about destination in advance. The challenge packet is just sent to the neighbour nodes. each neighbour communicated they got challenge to all neighbour with a specific end goal to decrease the convention overhead, challenge packet is communicated while challenge response is sent just to the source.[17]



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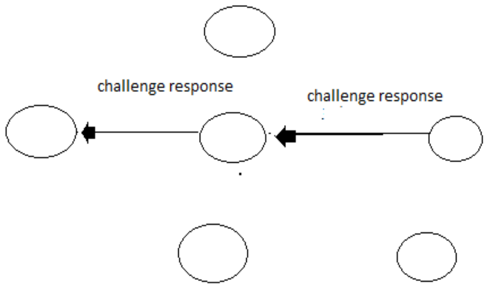
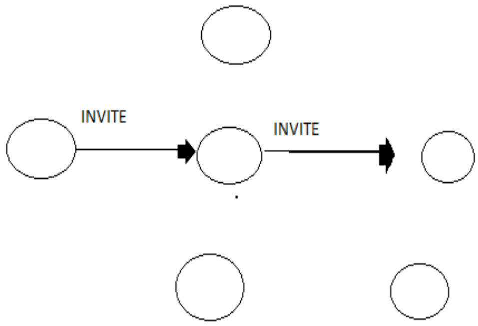
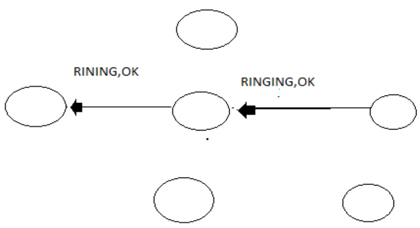


F ig 5.1An Example of a simple topology: the challenge phase.

Each node waits for small periods of time for the entire challenge packet. After the time is over, the node considered the best challenge packet.[19] Each challenging packet sent to each bidirectional node so all the neighbour node received the request and after some time challenge packet received by destination node (B).The destination node (B) again waited for some time and once time is over, node B collect all the challenging packet and reply to the best challenging packet. The path of to the lowest cost is used by node B to A.[17]



(a)



(b)

Fig 5.2 .An Example of a simple; the inviting phase.

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After the node A received the challenge response packet, it sends INVITE message having desired connection parameter to the node B. node B replied with OK and ringing message. In this way, a data screen set up from node A to node B after node A send the ACK message.[19] When a source started an SIP session, it sends a request for all the destination node wireless and each intermediate node stored the information as per their bandwidth availability. Once the destination node received the request for all possible path it choose the best one and response the choose path. [18]

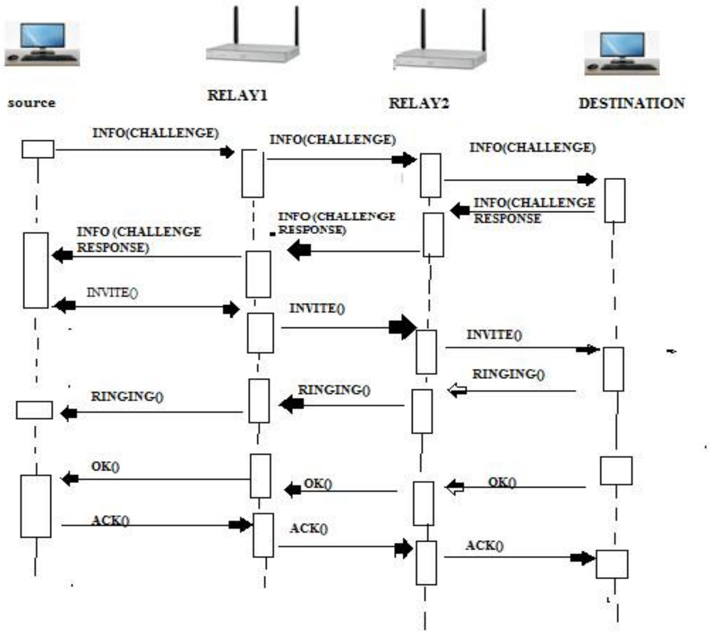

Fig 5.3An Example of a simple topology: the inviting phase

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ISSN (Print) : 2320-9798

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Vol. 6, Issue 3, March 2018

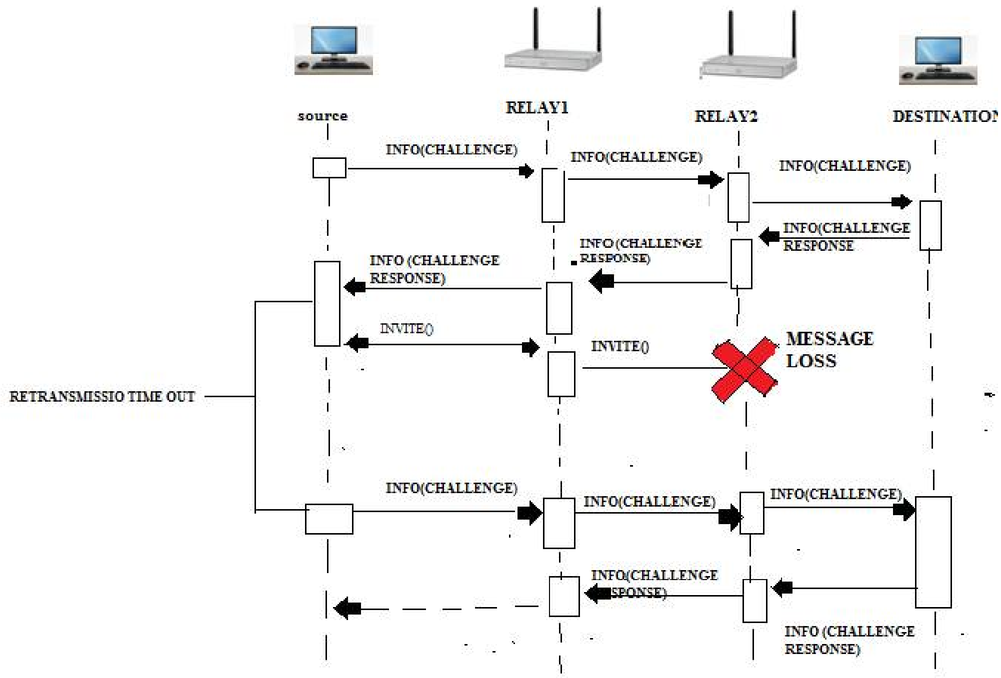

Fig 5.4An Example of a simple topology: the inviting phase, in the case of a message loss.

VI. SIP OVER H.323 PROTOCOL

SIP protocol depends on the current internet protocol and is used to set up and control the media session of the application layer protocol. It not just help unicast and multicast traffic, name mapping and redirection services, yet in addition help the execution of telecom services,[13] for example, call sending and call rejecting. Right now the broadly accepted VOIP control signalling advancements include H.323 standard of the ITU-T and SIP protocol of the internet Engineering Task Force IETF. The signalling control elements of the two advancements are essentially the same, however the plan style and execution techniques are unique.H.323 is an IP multimedia standard protocol suite which is designed based on broadcast communication arrange signalling and protocol. It provides a centralized and administration of the work mode.SIP using IP network protocols which provides multimedia service.[16]

Table 1. Comparison between H.323 and SIP

Standard	H.323	SIP
Organization	ITU	IETF
Expansibility	Poor	Good
Define area	Complete	Part
Simple Degree	Complex	Simple
Acceptance Level	More mainstream At present	Great development
Call routing	Gatekeeper	Location Server
Cost	High	Low
Core Server	Gatekeeper	Proxy/Register Server
Internet Fit	No	Yes

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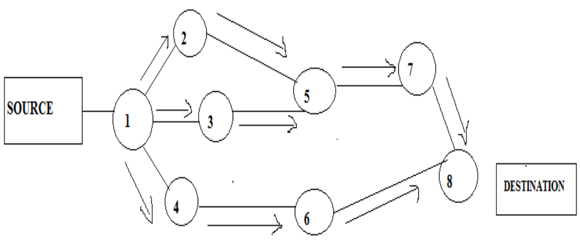
It can be seen from the similar examination in table 1 that SIP protocols inherit the characteristics of simple, open, and adoptable internet protocol. It is type of communication protocol which attracts high attention. Here, major areas of differences between H.323 and SIP. SIP is best for instant messaging, file sharing and multimedia communicating and other purpose for online game. On the other hand, H.323 is basically taken care of multimedia conferencing.[18]

VII. MANET ROUTING PROTOCOL

Routing Protocols assumes vital part in network for information transfer among nodes. At last, one intermediate node within the inter- network is essential through the process. The routing concept essential includes, two activity:- firstly, deciding right routing paths and secondly, exchanging the information among groups by the internetwork.
Ad –hoc routing protocol can be classified as-[6]
Proactive routing protocol are also called table driven. In this protocol it is obligatory for every one of the nodes to monitor route to all the conceivable destination nodes.
Reactive routing protocols are called demand on. In these protocol node find route when some other node proactive protocol a node has a least suspend. since at point a route is required for directing a packet .it is promptly chosen from the routing table where as reactive protocol has maximum delay because route are identifiable find on demand reactive protocol conclude low bandwidth then proactive protocol.[1][2]

VIII. AODV

AODV integrates benefit of both destination sequenced distance vector (DSDV) and dynamic source routing (DSR).[1] Ad hoc on demand distance is a reactive protocol. here nodes are generate route only when they are needed, under the AODV, the nodes just know their immediate neighbours and in this way entries on its routing tables cross pond just know to the neighbours nodes.[2] A node can monitor of its neighbour with HELLO packets that are transferred sequentially between among nodes. When source node sends a message to other nodes which is nit available to its neighbours then in this case AODV broadcast its message by source node with a specific end goal. The RREQ packets are read by whatever remains of the node to distinguish the goal of the packet, if a node knows a route to the goal, a Route Reply (RREP) message is sent to the source generally the Node will rebroadcast the RREP to its arrangement of neighbours. [13]



ROUTE REQUEST(RREQ)

ISSN(Online): 2320-9801
ISSN (Print) : 2320-9798

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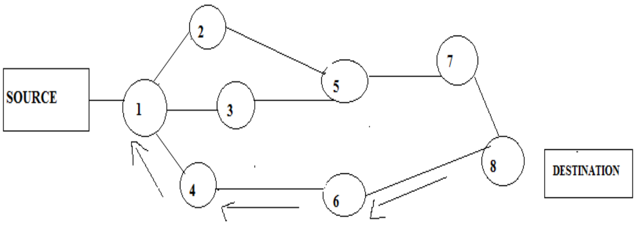
Vol. 6, Issue 3, March 2018



ROUTE REPLY(RREP)

IX. CONCLUSION

In this paper , we introduced the basic knowledge of VoIP over manet and reviewed the result recent work related to VoIP over manet. through study and analysis, we found there QOS of VoIP can be enhance by using such parameters voice mos, the end to end delay, voice jitter, throughput and packet delivery ratio.here we found that QOS enhance of different voice codes over ad hoc networks. here, we compared g.711, g.723 and g.729 three different codec on basis of performance. Through utilize parameters such as bit rate, framing, intervals, and payload packets. In this reviewed paper we found that VoIP decrease G.7231 is best codec in terms of providing an admissible MOS score. G.711 is not suitable for VoIP over ad hoc network in underground.

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