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# Performance Enhancement for QoS in VoIP Applications over MANET

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**Abstract:** *Mobile ad-hoc networking (MANET) has been intended as one of the most important technologies to stand future everywhere and invasive computing scenarios and internet connected mobile ad-hoc networking will be a necessary part of future wireless networks. For supporting multimedia services and Voice over IP (VoIP) in such situation, bear for Session Initiation Protocol (SIP) is necessary. A MANET is a decentralized collection of autonomous nodes but a SIP infrastructure requires centralized registrar servers and proxies. This paper proposes the implications of using standard SIP architecture in internet connected MANETs. This paper also examines performance limitations of SIP service measurable when centralized proxies/registrars located in access network when used by MANET nodes; then present and assess an alternative approach to provide SIP services in internet connected MANETs in order to regulate the impact of such performance limitations.*

**Keyword:** VoIP; SIP; MANET; MSIPMAN; QoS; UAS.

## 1. INTRODUCTION

Two major trends in technology can be observed more than the last decade: Internet and Pervasive communication. Pervasive communication is slowly but certainly becoming a vital part of our daily life. The Internet plays the vital role for everybody life, both in business, at home or during education. It is necessary for us to bridge those two technologies, where future wireless communication systems are projected to offer a wide range of multimedia services [1]. Mobile ad-hoc networking (MANETs) are reaching a stage where they can bear these services in order to provide an infrastructure helpful for the common user [2, 3]. Thus, the interconnection of MANETs to apply permanent infrastructure based IP networks is very significant in order to provide the ubiquitous user internet access everyplace at any time. The user inside an ad hoc network will get access to the public internet by using the packet forwarding abilities of transitional ad hoc network nodes towards the Access Router (AR) to the Internet, providing the user with access to different services through a variety of interconnected networks [4, 5].

This paper propose advanced services, such as multimedia communication for hybrid MANETs, maintain of Voice over IP (VoIP) service is necessary [6, 7]. A key issue in MANETs is session set up and management because, unlike in conventional networks, there is no centralized module to provide such a service. So far, session set up is essential to provide any appearance of communication beyond unpredictable, single message communication [8-10].

One significant aspect of VoIP technology is the Session Initiation Protocol (SIP), which is a signaling protocol for creating and controlling multimedia sessions. SIP is a client/server based protocol, and User Agent Server (UAS) needs to be available on the network to make SIP systems work correctly. As MA-

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NETs are dynamic networks created by peer nodes, standard SIP architecture has its problems as registrars and proxies are fixed, static and centralized, normally located in the access network [11-13].

This paper analyze and present the performance of two alternative approaches for SIP signaling, where the SIP proxy is co-located at the MANET gateway or located in the Access Network [14, 15]. The results show that using a proxy co-located at the gateway drastically reduces call setup latency for successful calls. And also, presents first results on capacity restrictions of voice traffic in such environment.

The rest of the paper is structured as follows, with Section 2 presents the main key problems related with the VoIP in MANET, dropping end-end relay, etc. Section 3 elaborates the multimedia communication in voice over IP over MANET. Section 4 explains the SIP architecture using OPNET. Section 5 explains the signaling flow of SIP message over multiple SIP servers. Section 6 explains the implementation of multiple SIP servers over MANET. Section 7 analysis the result and discussion part. Finally Section 8 concludes the performance enhancement over VoIP in MANET.

## 2. RELATED WORK

The main purpose of the VoIP system is to transport voice with the same quality as it is produced by the caller. Hence the key problem is VoIP system are end-end delay or latency, Delay Variance (Jitter), Packet Losses. Preventing voice packets from packet losses can be accomplished by providing an advanced queue management approach that monitors the initial congestion over the line and drops the incoming packets prior to the performance of VoIP [16, 17].

One more important issue is dropping end-end delay or latency. Latency is the time taken by the packet to accomplish from caller to callee. For each packet the latency or end-end delay is intended and it is compared with the limit life time of the packet to know whether the voice packet has its life time within sufficient limit or not. If the packet crosses its lifetime next it is discarded before accomplishment the callee without affecting the quality of VoIP packet as in [18-20].

OLSR (Optimized Link State Routing) is a proactive routing protocol which is a table driven protocol system that wills stores and updates the network routes. Whenever a route is needed, OLSR offers the route immediately without any initial delays of route exploration. In OLSR, some candidate nodes called multipoint relays (MPRs) are elected to have the accountability of forwarding the broadcast packets through traffic flooding process. This method reduces the overhead of packet transmission compared with the flooding mechanism. OLSR performs a hop by hop routing system, where every node uses its most updated routing information to route and deliver the packets.

SIP construction is based on centralized entities and two logical elements participates a key role in the architecture; registrar and proxy servers [21, 22]. Were as Registrars are the SIP entities where SIP users register their contact information once they connected to the SIP network. For example in an internet connected MANET registration scenario, a SIP user agent located within the MANET communicates to its registrar server situated in the Internet the SIP user name of the user(s) accessing the device, referred to as SIP Address of Records (AOR's) for that user, and the addresses where the user is accessible [23, 24]. That the Proxy servers can be co- located in the identical element as registrar server, are also needed since SIP users cannot identify, in most cases, the current full contact information of the callee but only its AOR. Therefore, SIP proxy server will acts as a user agents for forwarding or responding to the request received from another user agent or proxy [25].

The major problem of SIP signaling over MANET includes SIP user lookup time, the mobility aid of the terminal, and the interoperability between Internet users and SIP over MANET users [26].

Thus, the mechanism working as a fixed IP networks to locate the SIP proxy and to map the SIP user name with destination IP address involves centralized servers that could not exist in a MANET [27]. Hence, the SIP protocol cannot be deployed as in remote MANETs. In internet connected MANETs however, end points located in the ad-hoc network can achieve other parties located in the internet through gateway nodes connecting the MANET with the public internet, except when two MANET nodes need to communicate via SIP, any SIP signaling will pass through the gateway, which is a strict performance restriction. Therefore, an alternative approach is desirable [28].

## 3. PROPOSED VoIP OVER MANET

MANET provides an appropriate platform for the exploitation of VoIP applications in lots of scenarios as the quality of service necessities is a major challenge for real time applications. This segment provides a concise overview of VoIP implementation with voice codec's, H.323 signaling protocol [29].

VoIP-Now a day's multimedia communication VoIP technology is most suitable that allows communication over packet switched network among two parties. It is a tactics that performs the transmission of voice pattern and multimedia sessions over Internet that minimizes the cost of extensive distance voice calls or sometimes offers with free of cost service irrespective of the distance. Initially human voice leftovers in the analog form so prior to transmitting signals this voice over packet has to be switched from analog signal to digitize to the sender side and the

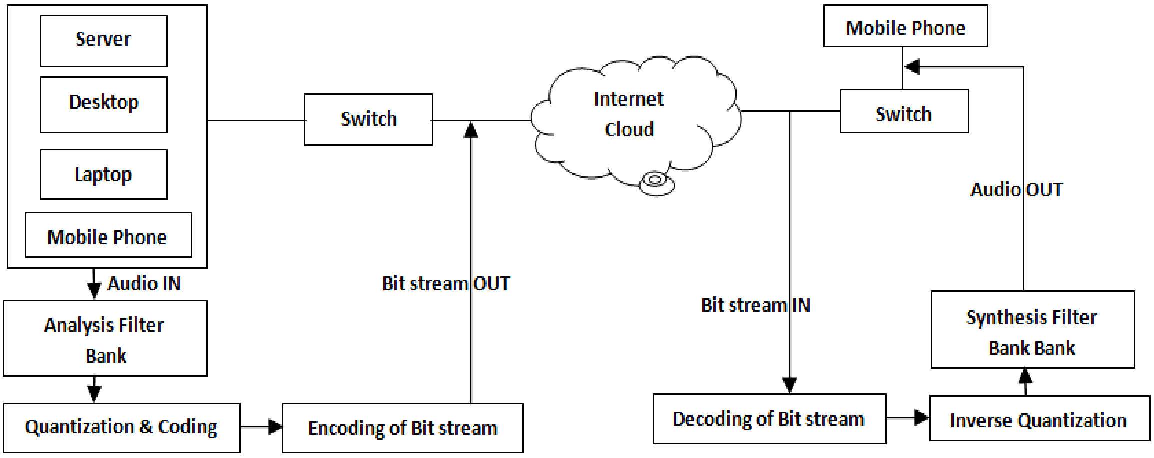


*Figure 1 Voice over IP over MANET*

reverse process is performed by the receiver end. Digitization process is composed of sampling, quantization and encoding. The quality of voice communication over IP is influenced by the choice of voice codec. Voice communication in a VoIP system can be demonstrated by a block diagram as shown in Figure 1.

a) *CODECs*- The purpose of voice codec process is the early step for voice communication. Its main function is to translating the incoming analog voice signal into digital stream. International Telecommunication Unions (ITU-T) has developed different encoding techniques, which are classified as waveform coders (eg-G.711, G.726, etc) [30].

Vocoders and Hybrid Coders (eg- G.728, G.723, G.729). As the number of mobile nodes varies then the quality of the voice also gets affected. Due to this choice of codec is the significant factor whose main purpose is to perform voice digitization and compression the signal will ensuring the lowest bit rate possible degrading the signal quality [31, 32].

b) *Signaling protocol*- The encoded speech in the packets is when transmitted over IP network requires suitable signaling protocol like H.323 or SIP. H.323 is a ITU-Ts suggested standard protocol for set up and scratch losing calls of IP telephony. H.323 is responsible for encoding, packetizing, decoding, signaling of audio and video signals [33].

## 4. SIP ARCHITECTURE

SIP is an application-layer control protocol that can generate, alter, and conclude multimedia sessions such as Internet telephony calls. Session Initiation Protocol can also invite participants to already-existing sessions, such as multicast conferences. The Media can be added as an accessible session. SIP obviously supports redirection services and name mapping services, which supports personal mobility.

The message type of SIP is Request-Response message as shown in Figure 2 [34]. However, in SIP simulation a third scenario which contains a new SIP topology is created.

The main key entities in SIP are:

**Proxy Server** - Proxy Server is an intermediary entity that acts as both a server and a client for the function of making requests on behalf of other clients.

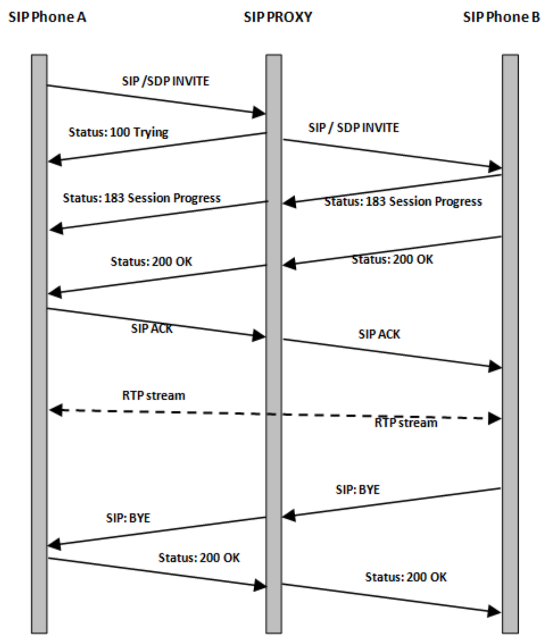


*Figure 2 The message type of SIP*

**User Agent** - User Agent is the endpoint entity. It initiates and terminates sessions by exchanging requests and response.

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**Registrar** - Registrar is a server that updates location information into the database with the contact information of the user.

**Redirect Server** - Redirect Server is a server that maps the SIP address of the called party into zero or new addresses and returns them to the client.

## 5. MULTIPLE SERVERS BASED SIP SYSTEM

The multiple server SIP is based on the client /server architecture in which all the servers, such as proxy server, location server, and registration server, respond to the request sent by the UAC separately. The UAC caller registers itself with the Registration server by sending a 'REGISTER' message. After receiving the 'REGISTER' message from the registration server, it extracts the user name, IP address, and port number then stores them in the location server. A contact header field of the 'REGISTER' message grasps information on the lifespan of the registration. Likewise, the UAS callee also registers itself at the registration server. The location details of both the UAC caller and the UAS callee are stored in the location server. An 'INVITE' message is sent by the UAC callee to the redirect server. Multiple servers use the Redirect server for initiating a session between a UAC caller and a UAS callee. The Redirect server does not forward the request on behalf of the UAC; it only returns the location shown in Figure 3 [34].

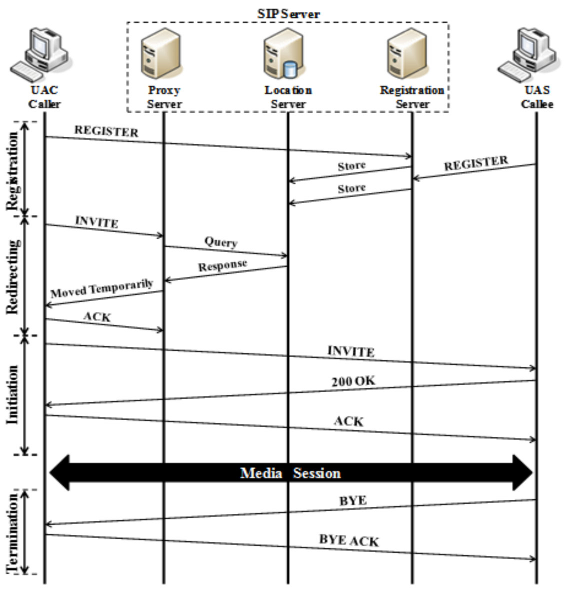


*Figure 3 Signaling flows of SIP messages over multiple SIP servers*

The 'INVITE' message consists of the header fields, such as 'INVITE', Via, Max-Fwd, To, From, Call-ID, CSeq, Subject, Contact-type, and Content-length. The Redirect server performs a look-up inside the database of the location server for the proposed recipient. Then the user position information will be sent back to the UAC in a redirection class response.

The response Moved Temporarily contains the message format having header fields SIP moved temporarily, Via, To, From, Call-ID, CSeq, Contact- type and Contact-length. After getting the response, the UAC callee acknowledges using an ACK response. At this stage, the redirection process and the exchange process are completed. A new INVITE message is sent directly to the UAS callee as the location is obtained from the control header field of Moved Temporarily in response to the redirect server. The new INVITE message contains a new Call-ID.

In reply to the 'INVITE' message, a direct 200 'OK' reply is sent instead of the 180 ringing response. The UAC caller responds to the UAS callee by acknowledging it using an 'ACK' reply. Thus, a session is initiated among the UAC caller and the UAS callee using a redirect server. After initiating the session, the media session is in progress between the UAC caller and the UAS callee using the RTP protocol. Once the media session is completed, the session is terminated by sending a 'BYE' request.

Once it is acknowledged by the UAC caller, the complete session is terminated. When using multiple servers based SIP, the redirect server doesn't forward session initiation requests for the UAC caller as is done by the proxy server. Because the redirect server does not initiate the request, a lower state overhead is desired compared to a proxy server. Multiple server based SIP uses the redirect server which processes very few messages, therefore it has high processing capacity.

## 6. IMPLEMENTATION OF MULTIPLE SIP SERVERS OVER MANET

We propose a distributed protocol called MSIPMAN that permit SIP implementation in MANETs. This protocol is evaluated on the Network Simulator-2 (NS-2) where evaluation is made with the Tightly Coupled Approach using metrics such as average session establishment time, failure rate and consumed bandwidth. The evaluation explains that the projected protocol in has enhanced performance in terms of adaptability and scalability to node mobility.

The projected solution in chooses a group of nodes that are mobile to act as SIP servers, and they establish a virtual infrastructure as cover on top of the physical network. A distributed algorithm is built to assemble the topology and to assign dynamically previously explained functionality to a cluster of nodes in the network. The replication results obtained using the NS-2 simulator clearly show that the proposed MSIPMAN protocol is well-adapted to mobile ad hoc network. MSIPMAN has a low control overhead, lower session establishment time and high service

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availability.

Separate from the results obtained using the NS-2 simulation tool; this job has not been confirmed using real results that could be obtained in a real life scenario. Proactive route optimization in SIP mobility is introduced in. The author's motivation for this work is to attain latency reduction in session setup.

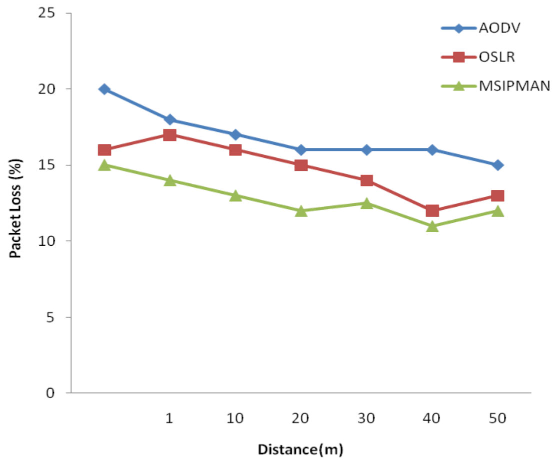


*Figure 4 Packet Loss Results*

The above Figure 4 shows the packet loss result based on the packet loss percentage and by the distance travelled which are managed by the SIP server. The SIP server was used to initiate, terminate and modify the SIP calls. The more number of VoIP calls that SIP server can support, the better performance it can provide.

In the proposed method, the mobility binding information is pre-fetched and used for session establishment during the location registration step. Using the proactive route optimization, reduced latency in session setup is achieved by eliminating the traversal over multiple SIP servers. Whenever a session is initiated, the session will be establishment directly with the callee is possible if the caller has suitable mobility binding information.

A mobility-aware pre-fetching method is developed here only the lower mobility binding information is chosen because it is most likely that such information could be used for session establishment. Also in, the authors recommend a new session setup procedure where mobility information with enough residual time is used. This work lacks wide simulations using the developed analytical models in order to verify the projected procedures and optimization level accomplished.

## 7. RESULT AND DISCUSSION

The existing performance improvement methods for SIP- based applications over MANET vary in terms of system features, requirements, feasibility in implementation, incorporation with presented systems, and costs. In common, the key performance improvement methods are related to the dynamic adjustments for SIP timers, dynamic adjustments for the routing protocol parameters, implementations for supportive signaling systems, infrastructural based solutions, or examine distribution features for the system users.

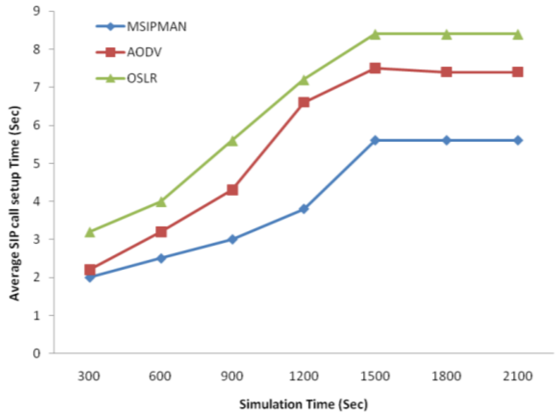


*Figure 5 Average SIP call setup time in seconds*

The above Figure 5 shows the average SIP call setup time in seconds for VoIP applications over different mobility models.

The energetic adjustments for SIP timers present flexible implementation for SIP-based applications over diverse platforms. This assessment relates to imaginary studies, in reality still, the SIP adjustments need to think about the nature of the network systems that SIP signaling is working on.

Thus, applying the dynamic adjustments for SIP signaling systems is not a proper solution which can be applied over MANET systems unless the nature of MANET systems had considered this method. The wireless and mobility characteristics of MANET affect the SIP signaling performance.

This method is measured as one of the most effective performance augmentation methods. But, no efficient level of implementations has been proven for this method, especially for SIP-based VoIP over MANET for emergency and backup scenarios. The implementation for caring signaling systems for SIP is considered as one of the effective solutions.

Though, the lossy nature of MANET is also affects the performance of SDP (Session Description Protocol) which increases the performance problems of SIP signaling. Most research studies in the literature implement SIP without SDP. So, the SDP signaling system improves the SIP signaling performance over MANET as it supports the management features of the SIP signaling system.

The Figure 6 shows the delay effect is inversely proportional to that of the number of mobile nodes to

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the quality on the VoIP calls based on the AODV, OSLR and MSIPMAN routing protocols for IPv4.

During a SIP based VoIP call initialization, when a caller dials a number on a SIP phone, a SDP message is attached to the SIP INVITE message which is sent to the IP PBX the SIP phone is registered to. In the SDP message, connection details, media details and DTMF event types are advertised.

Typically, such information is sent from the caller's SIP phone to the IP PBX, which is then relayed to the other SIP phone which is receiving the call. The SIP phone receiving the call which at this stages it is still being established, also sends SDP data back to the IP PBX which is relayed to the SIP phone making the call. Because of such process, if the call is established the SIP phones taking part in this SIP based VoIP call know to where the media stream should be sent and what type of media and codec to use. They also now know what media type and codec they will be receiving.

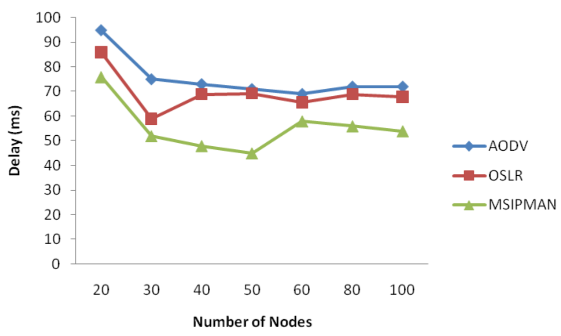


*Figure 6 The delay effect is inversely proportional to that of the number of mobile nodes to the quality of the VoIP calls*

On the other side, the dynamic adjustments for the parameters of MANET routing protocols have exposed an efficient enhancement for different implemented applications. This method depends on obliging the routing parameters to provide the best level of service for the implemented applications. SIP-based applications using this method show an enhanced level in performance for the SIP signaling and voice data transfer in general.

Synchronization issues between SIP and SDP protocols has been apprehension especially the performance of SIP signaling in network systems that are variable in nature or mobility linked in their implementation. The infrastructural based solutions use methods to improve the SIP implementations over MANET. One of the suggested methods equipment multiple SIP servers with high performance in order to support larger numbers of MANET nodes. Though, this method is difficult to implement for emergency or communication backup scenarios because of the required synchronization functionality among multiple SIP servers for the mobile callers.

From the Result, it represent that the Merging together dynamic methods for SIP and MANET has a shows potential level of performance improvement with lower costs and simple implementation. But, this improvement method needs to be based on the evaluation studies for the current state-of-the-art for SIP signaling over different MANET scenarios. In addition, the implementation of these improvement methods has not been fully investigated over clearly recognized mobility models for MANET nodes. The simulation or test-bed tools used do not mirror reliable results that can be considered as reference results for the investigated methods.

## 8. CONCLUSION

This evaluation study used to evaluate the main performance metrics for SIP based VoIP applications with MANET routing protocols using different mobility models. Despite of the QoS issues, without a central SIP server, it will be tricky to communicate with a large number of MANET-based callers. Additional infrastructural methods suggested in the literature include controlling the speed of nodes, limiting the hop numbers, and dropping the background traffic of other simultaneous applications.

Other research efforts suggest the use of service distribution features over the system users by scheduling the calls setup processes. The key purpose of these methods is to reduce concurrent calls by applying the time distribution features over the service users to increase the QoS level for the provided services. The advantage and disadvantage of the reviewed performance enhancement methods vary in terms of the development level and completion requirements. However, both (SIP parameters and MANET parameters) will do the dynamic adjustment for methods show a good level of performance enhancement. Thus, the most efficient method for enhancing SIP signaling performance over MANET is to qualify the SIP signaling behavior to conform to the mechanical nature of MANET systems.

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## Authors Biography

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