VOICE COMMUNICATION OVER MOBILE AD-HOC NETWORKS

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Certificate

This is to certify that this thesis is the outcome of thorough study and research done by **Nafees Uddin Ahmed** under the supervision of **Dr. Md. Humayun Kabir**, **Associate Professor** of **Computer Science and Engineering** at **Bangladesh University of Engineering and Technology**. This report is composed by the undersigned and it is not a reproduction of any report submitted elsewhere. This thesis is hereby declared to be self-contained and without any resemblance with any other report of this sort.

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Abstract

The purpose of this thesis is to study different issues concerning voice communication over mobile ad-hoc networks and try to find some solves for them. Starting with basic understanding of ad-hoc networks, its applicability and somewhat details of existing works of ad-hoc routing, this thesis digs into different troubles we face in ad-hoc voice communication. After that the research work focuses onto betterment of SIP Architecture support in ad-hoc realm. A new protocol is proposed in this purpose to improve performance over existing works and details is given on how the protocol works and how it would behave in different situations. A section is also provided on how to set up the simulation environment to analyze performance of such protocol. But in order to maintain brevity and conciseness, their discussion has been limited.

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Chapter 1

Introduction

1.1 Motivation

Whenever there is an opportunity of achieving some services or goods for free, it's a common human response is to get hold of it even if there is a compromise to be made about quality, performance or security. In modern world the burden of distant communication is mostly handled by cellular networks, which is popular, reliable, easily available, widely accepted but comes at a price. Now if someone provides the mass people with something that can enable them to skip the existing cellular communication structure and have voice conversation for hours without having to worry about credits running out, surely, it would be very interesting. In a perfect world it might have been possible to provide all possible communications for free. But in reality we can provide such facility in some small scale. When a caller tries to call someone from cellular phone, who is geographically very close to him, we might think it as a waste of cellular network because we are trying to make a communication through complicated cellular architecture that might have been really easily made directly between two devices. Since every cellular phone is some kind of sophisticated radio device and can connect to a base station that is far off, it already has the capability of connecting to another device that is nearby. So if we can somehow provide a way of connecting those two devices without the help of any central architecture, we can allow free, independent communication between caller and callee.

The idea of communicating without the help of any central architecture is known as ad-hoc communication. Ad-hoc network is a general concept and can be created on any kind of radio capable network mechanism. Most suitable of them is wireless LAN or (1)WiFi. With more and more devices being made WiFi enabled (including cell phones), having ad-hoc networks available is becoming easier day by day. Besides the obvious and elusive feature of free calling, having facility of voice communication over such a network can provide us a lot of advantages in some situations with deferent perspectives. The base concept of such communication is to avoid any infrastructure based communication. So whenever we face a situation where the reliability of such structures comes into question, we have our system come to rescue.

- Every infrastructure based voice communication system (GSM, CDMA etc) is driven by centralized structures like Base Station/MTSO etc., which are limited in capacity and can only provide for some statistically average load. When a huge gathering happens, like in a football match or concert or some procession they miserably fail to support voice communication. In such an extreme cases, ad-hoc network can provide useful alternate way of communication between people.

- In case of natural calamity like earthquake, flood, cyclone infrastructure based communication is bound to suffer disturbance or lack of operability. Rescue or relief teams generally have to be well equipped with expensive equipments to enable communication between individuals. Introduction of ad-hoc network based voice communication can serve well in this purpose. Also people suffering such disaster can have a way to communicate with each other and the rescue team. Since this network doesn't require any predefined setup or any non-regular device, connectivity is instant and useful.

- When communication between groups is required to be secured (like covert teams) avoidance of public network is necessary. Ad-hoc network in some secured channel with proper encryption can go a long way here in providing some solution.

So voice communication over such alternate network can be really handy.

1.2 Problem Specification

To transmit real time data like voice over any network, there are always some common requirements to meet to bring out acceptable performance. To support voice transmission a network should provide,

- Minimal delay in connection setup
- Minimal delay in packet transmission
- Minimal jitter
- Minimal hand-off time
- Adequate bandwidth allocation to transmit voice packets etc.

When we think about voice transmission, we need to think about these properties of a network. There had been some real progress in voice transmission over IP networks. But in case of ad-hoc networks it's still in the early stages of research cycle.

Like any other infrastructureless network, ad-hoc network suffers from lot of complexities due to its dynamic nature and it fails to provide the reliability of a structured network. There are lots of things to be done in this field. The primary goal of this research work is to go in deep of this field and to make ad-hoc voice communication as close as possible to voice communication systems available to general users in terms of both performance and usability.

Existing ad-hoc structure provides us with dynamic mechanism of routing and IP assignment of nodes, and gives the upper layer a way of sending datagram packets from one node to another node. But the service is best of effort without any promise of QoS. Aside from assurance of voice transmission quality, we have few more issues to consider when we try to match our system existing well-developed cellular network. One issue is subscriber identification. When we try to mail someone, we use a mail address. When we try to call someone we use a phone number that is unique to that person. In case of ad-hoc network, when we start up some device enabled with ad-hoc networking capability, what we are presented with is a list of IP addresses or MAC addresses. This might be meaningful to lot of researchers but when general communication is concerned, this is just meaningless. Since nodes are assigned with dynamic IP addresses we can never tell which one is who, just by looking at such list. To support usability of such network, we must provide with something like email address or phone number that can identify a user among all these devices.

Now there are many existing mechanism of translating some human identifiable name into IP address or other number (one being DNS). In recent times SIP (session initiation protocol) has gained a lot of credibility in the realm of VoIP for its simplicity and acceptability. SIP also provides a useful mechanism similar to DNS in translating SIP URI into IP address. But the architecture of SIP is built assuming that it would be implemented on an infrastructure-based network. But in case of ad-hoc network this is quite impossible to have some special node that can act like some server and be dedicated. So to allow users to call others using some name or number we need to adopt SIP into dynamic non-centralized structure of ad-hoc network. This thesis work is wholly devoted to this problem.

1.3 Related Works

There had been some effort made to adopt SIP architecture onto ad-hoc networks (2) (3) (4) (5) . But most of them lacked the required performance criteria. Few of those tried to create an overlay network on top of ad-hoc network to give a common naming mechanism. But maintaining such a network can be costly in terms of overhead generated. Such methods require a lot of broadcasts at the application level. Since the underlying routing mechanism itself does a lot of flooding there is already some control overhead. When addition overhead is generated on its top, quality of communication degrades a lot due to lack of available bandwidth. One way to subside this problem is to merge SIP architecture with routing broadcast. Merging network layer to optimize performance might seem irrational at first but the performance advantage can be well worth.

Li Li and Louise Lamont published a paper (4) with a view to utilize optimized flooding mechanisms provided by underlying routing algorithm and support SIP registration service to SIP nodes available in the network. Here the base concept is that every SIP node is also a registrar server itself. Each of them saves a table with URI to IP address mapping. Each SIP node periodically broadcasts register message in the network. Any other node receiving that message updates its table. Whenever it needs to communicate with someone else on the network, it just looks up the table and connects using the IP address. Problem with this mechanism is there is still broadcast from each node. The paper states that if there is some registrar server available on the network, SIP nodes just stops broadcasting on their part and registers themselves on that registrar server. But they don't make any kind of reference about how the server was created or what happens when they goes down. If we can manage such registrar server obviously control overhead of maintaining SIP architecture decreases, but again we face some other problems. Every time a node tries to connect to other node on the network it has to make a query to the registrar server and waits for response. So data packet concentration around that server increases by a huge amount and creates congestion. Having a single registrar server also creates the problem of single point of failure. When this node goes down the whole network will go down. So existing works provide us with two solutions to adopt SIP, one is totally decentralized with lot of broadcast and the other one with totally centralized mechanism with possible congestion and downtime. In our work, we can try to make something better by utilizing both the concept and creating something that lies in between.

1.4 Project Organization

This thesis dissertation describes the design and performance measure of the protocol to incorporate SIP architecture onto ad-hoc networks.

Chapter 2 provides background reading required to understand the protocol developed.

Chapter 3 specifies in details how the protocol works.

Chapter 4 gives an overview on performance testing of the protocol in simulated environment. It also provides details on how to setup such simulation.

Chapter 5 will suggest improvements and extensions for future development and the thesis will conclude with a brief summary.

Chapter 2

General Concepts

2.1 Wireless ad-hoc networks

2.1.1 General

A wireless ad-hoc network (6) is a collection of mobile/semi-mobile nodes with no pre-established infrastructure, forming a temporary network. Each of the nodes has a wireless interface and communicates with each other over either radio or infrared. Laptop computers and personal digital assistants that communicate directly with each other are some examples of nodes in an ad-hoc network. Nodes in the ad-hoc network are often mobile, but can also consist of stationary nodes, such as access points to the Internet. Semi mobile nodes can be used to deploy relay points in areas where relay points might be needed temporarily.

Figure 1 shows a simple ad-hoc network with three nodes. The outermost nodes are not within transmitter range of each other. However the middle node can be used to forward packets between the outermost nodes. The middle node is acting as a router and the three nodes have formed an ad-hoc network.

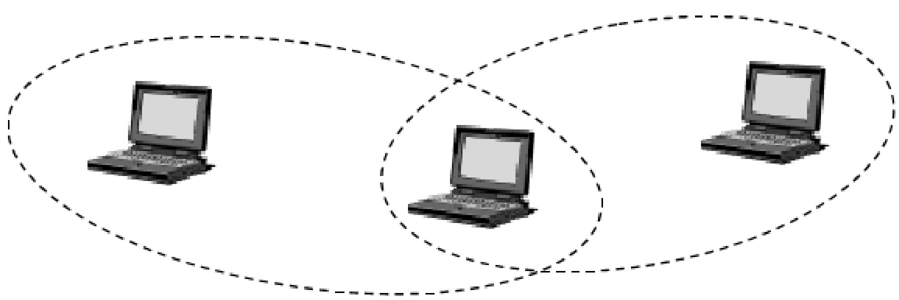


Figure 1 Example of a three node mobile ad-hoc network

An ad-hoc network uses no centralized administration. This is to be sure that the network won't collapse just because one of the mobile nodes moves out of transmitter range of the others. Nodes should be able to enter/leave the network as they wish. Because of the limited transmitter range of the nodes, multiple hops may be needed to reach other nodes. Every node wishing to participate in an ad-hoc network must be willing to forward packets for other nodes. Thus every node acts both as a host and as a router. A node can be viewed as an abstract entity consisting of a router and a set of affiliated mobile hosts (Figure 2). A router is an entity, which, among other things runs a routing protocol. A mobile host is simply an IP-addressable host/entity in the traditional sense. Ad-hoc networks are also capable of handling topology changes and malfunctions in nodes. It is fixed through network reconfiguration. For instance, if a node leaves the network and causes link breakages, affected nodes can easily request new routes and the problem will be solved. This will slightly increase the delay, but the network will still be operational.

Wireless ad-hoc networks take advantage of the nature of the wireless communication medium. In other words, in a wired network the physical cabling is done a priori restricting the connection topology of the nodes. This restriction is not present in the wireless domain and, provided that two nodes are within transmitter range of each other, an instantaneous link between them may form.

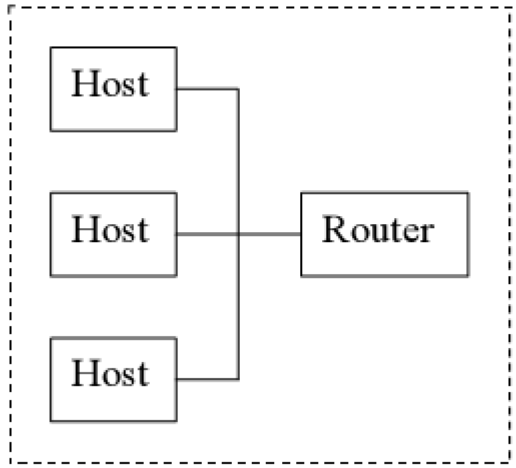


Figure 2 Block diagram of mobile node acting both as hosts and as router

2.1.2 Usage

There is no clear picture of what these kinds of networks will be used for. The suggestions vary from document sharing at conferences to infrastructure enhancements and military applications. In areas where no infrastructure such as the Internet is available an ad-hoc network could be used by a group of wireless mobile hosts. This can be the case in areas where a network infrastructure may be undesirable due to reasons such as cost or convenience. Examples of such situations include disaster recovery personnel or military troops in cases where the normal infrastructure is either unavailable or destroyed.

Other examples include business associates wishing to share files in an airport terminal, or a class of students needing to interact during a lecture. If each mobile host wishing to communicate is equipped with a wireless local area network interface, the group of mobile hosts may form an ad-hoc network. Access to the

Internet and access to resources in networks such as printers are features that probably also will be supported.

With more and more portable devices with WiFi radio pre-built coming into market soon it might be get some real acceptance in the filed of insecure but free voice communication over short rage in case of campus talk or disaster scenarios where no infrastructure might be up and running.

2.1.3 Characteristics

Ad-hoc networks are often characterized by a dynamic topology due to the fact that nodes change their physical location by moving around. This favors routing protocols that dynamically discover routes over conventional routing algorithms like distant vector and link state (7). Another characteristic is that a host/node have very limited CPU capacity, storage capacity, battery power and bandwidth, also referred to as a "thin client". This means that the power usage must be limited thus leading to a limited transmitter range. The access media, the radio environment, also has special characteristics that must be considered when designing protocols for ad-hoc networks. One example of this may be unidirectional links. These links arise when for example two nodes have different strength on their transmitters, allowing only one of the host to hear the other, but can also arise from disturbances from the surroundings. Multihop in a radio environment may result in an overall transmit capacity gain and power gain, due to the squared relation between coverage and required output power. By using multihop, nodes can transmit the packets with a much lower output power.

2.2 Routing

Because of the fact that it may be necessary to hop several hops (multi-hop) before a packet reaches the destination, a routing protocol is needed. The routing protocol has two main functions, selection of routes for various source-destination pairs and the delivery of messages to their correct destination. The second function is conceptually straightforward using a variety of protocols and data structures (routing tables).

2.2.1 Conventional protocols

If a routing protocol is needed, why not use a conventional routing protocol like link state or distance vector? They are well tested and most computer communications people are familiar with them. The main problem with link-state and distance vector is that they are designed for a static topology, which means that they would have problems to converge to a steady state in an ad-hoc network with a very frequently changing topology.

Link state and distance vector would probably work very well in an ad-hoc network with low mobility, i.e. a network where the topology is not changing very often. The problem that still remains is that link-state and distance-vector are highly dependent on periodic control messages. As the number of network nodes can be large, the potential number of destinations is also large. This requires large and frequent exchange of data among the network nodes. This is in contradiction with the fact that all updates in a wireless interconnected ad hoc network are transmitted over the air and thus are costly in resources such as bandwidth, battery power and CPU. Because both link-state and distance vector tries to maintain routes to all reachable destinations, it is necessary to maintain these routes and this also wastes resources for the same reason as above.

Another characteristic for conventional protocols are that they assume bi-directional links, e.g. that the transmission between two hosts works equally well in both directions. In the wireless radio environment this is not always the case.

Because many of the proposed ad-hoc routing protocols have a traditional routing protocol as underlying algorithm, it is necessary to understand the basic operation for conventional protocols like distance vector, link state and source routing.

2.2.2 Link State

In link-state routing (7), each node maintains a view of the complete topology with a cost for each link. To keep these costs consistent; each node periodically broadcasts the link costs of its outgoing links to all other nodes using flooding. As each node receives this information, it updates its view of the network and applies a shortest path algorithm to choose the next-hop for each destination.

Some link costs in a node view can be incorrect because of long propagation delays, partitioned networks, etc. Such inconsistent network topology views can lead to formation of routing-loops. These loops are however short-lived, because they disappear in the time it takes a message to traverse the diameter of the network.

2.2.3 Distance Vector

In distance vector (7) each node only monitors the cost of its outgoing links, but instead of broadcasting this information to all nodes, it periodically broadcasts to each of its neighbors an estimate of the shortest distance to every other node in the network. The receiving nodes then use this information to recalculate the routing tables, by using a shortest path algorithm. Compared to link-state, distance vector is more computation efficient, easier to implement and requires much less storage space. However, it is well known that distance vector can cause the formation of both short-lived and long-lived routing loops. The primary cause for this is that the nodes choose their next-hops in a completely distributed manner based on information that can be stale.

2.2.4 Source Routing

Source routing (7)means that each packet must carry the complete path that the packet should take through the network. The routing decision is therefore made at the source. The advantage with this approach is that it is very easy to avoid routing loops. The disadvantage is that each packet requires a slight overhead.

2.2.5 Flooding

Many routing protocols uses broadcast to distribute control information, that is, send the control information from an origin node to all other nodes. A widely used form of broadcasting is flooding (7) and operates as follows. The origin node sends its information to its neighbors (in the wireless case, this means all nodes that are within transmitter range). The neighbors relay it to their neighbors and so on, until the packet has reached all nodes in the network. A node will only relay a packet once and to ensure this some sort of sequence number can be used. This sequence number is increased for each new packet a node sends.

2.2.6 Classification

Routing protocols can be classified (8) into different categories depending on their properties.

- Centralized vs. Distributed
- Static vs. Adaptive
- Reactive vs. Proactive

One way to categorize the routing protocols is to divide them into centralized and distributed algorithms. In centralized algorithms, all route choices are made at a central node, while in distributed algorithms, the computation of routes is shared among the network nodes.

Another classification of routing protocols relates to whether they change routes in response to the traffic input patterns. In static algorithms, the route used by source-destination pairs is fixed regardless of traffic conditions. It can only change in response to a node or link failure. This type of algorithm cannot achieve high throughput under a broad variety of traffic input patterns. Most major packet networks uses some form of adaptive routing where the routes used to route between source-destination pairs may change in response to congestion

A third classification that is more related to ad-hoc networks is to classify the routing algorithms as either proactive or reactive. Proactive protocols attempt to continuously evaluate the routes within the network, so that when a packet needs to be forwarded, the route is already known and can be immediately used. The family of Distance-Vector protocols is an example of a proactive scheme. Reactive protocols, on the other hand, invoke a route determination procedure on demand only. Thus, when a route is needed, some sort of global search procedure is employed. The family of classical flooding algorithms belongs to the reactive group. Proactive schemes have the advantage that when a route is needed, the delay before actual packets can be sent is very small. On the other side proactive schemes needs time to converge to a steady state. This can cause problems if the topology is changing frequently.

2.3 Ad-hoc routing protocols

2.3.1 Desirable properties

If the conventional routing protocols do not meet our demands, we need a new routing protocol. The question is what properties such protocols should have? These are some of the properties (9) that are desirable:

Distributed operation: The protocol should of course be distributed. It should not be dependent on a centralized controlling node. This is the case even for stationary networks. The difference is that nodes in an ad-hoc network can enter/leave the network very easily and because of mobility the network can be partitioned.

Loop free: To improve the overall performance, we want the routing protocol to guarantee that the routes supplied are loop-free. This avoids any waste of bandwidth or CPU consumption.

Demand based operation: To minimize the control overhead in the network and thus not wasting network resources more than necessary, the protocol should be reactive. This means that the protocol should only react when needed and that the protocol should not periodically broadcast control information.

Unidirectional link support: The radio environment can cause the formation of unidirectional links. Utilization of these links and not only the bi-directional links improves the routing protocol performance.

Security: The radio environment is especially vulnerable to impersonation attacks, so to ensure the wanted behavior from the routing protocol, we need some sort of preventive security measures. Authentication and encryption is probably the way to go and the problem here lies within distributing keys among the nodes in the ad-hoc network. There are also discussions about using IP-sec (10)that uses tunneling to transport all packets.

Power conservation: The nodes in an ad-hoc network can be laptops and thin clients, such as PDAs that are very limited in battery power and therefore uses some sort of stand-by mode to save power. It is therefore important that the routing protocol has support for these sleep-modes.

Multiple routes: To reduce the number of reactions to topological changes and congestion multiple routes could be used. If one route has become invalid, it is possible that another stored route could still be valid and thus saving the routing protocol from initiating another route discovery procedure.

Quality of service support: Some sort of Quality of Service support is probably necessary to incorporate into the routing protocol. This has a lot to do with what these networks will be used for. It could for instance be real-time traffic support.

None of the proposed protocols from MANET have all these properties, but it is necessary to remember that the protocols are still under development and are probably extended with more functionality. The primary function is still to find a route to the destination, not to find the best/optimal/shortest-path route.

2.3.2 MANET

IETF has a working group named MANET (Mobile Ad-hoc Networks) (11) that is working in the field of ad-hoc networks.. Even if MANET currently is working on routing protocols, it also serves as a meeting place and forum, so people can discuss issues concerning ad-hoc networks. Currently they have seven routing protocol drafts:

- AODV - Ad-hoc On Demand Distance Vector (12)
- ZRP - Zone Routing Protocol (13)
- TORA / IMEP - Temporally Ordered Routing Algorithm / Internet MANET Encapsulation Protocol
- DSR - Dynamic Source Routing (14) (15)
- CBRP - Cluster Based Routing Protocol (16)
- CEDAR - Core Extraction Distributed Ad hoc Routing (17)
- AMRoute – Ad-hoc Multicast Routing Protocol (18)
- OLSR - Optimized Link State Routing Protocol (19)

Of these proposed protocols we have chosen to analyze few of those with OLSR getting a little bit more attention.

2.3.3 Destination Sequenced Distance Vector - DSDV

Description

DSDV (20) is a hop-by-hop distance vector routing protocol that in each node has a routing table that for all reachable destinations stores the next-hop and number of hops for that destination. Like distance-vector, DSDV requires that each node periodically broadcast routing updates. The advantage with DSDV over traditional distance vector protocols is that DSDV guarantees loop-freedom.

To guarantee loop-freedom DSDV uses a sequence numbers to tag each route. The sequence number shows the freshness of a route and routes with higher sequence numbers are favorable. A route R is considered more favorable than R' if R has a greater sequence number or, if the routes have the same sequence number but R has lower hop-count. The sequence number is increased when a node A detects that a route to a destination D has broken. So the next time node A advertises its routes, it will advertise the route to D with an infinite hop-count and a sequence number that is larger than before. DSDV basically is distance vector with small adjustments to make it better suited for ad-hoc networks.

These adjustments consist of triggered updates that will take care of topology changes in the time between broadcasts. To reduce the amount of information in these packets there are two types of update messages defined: full and incremental dump. The full dump carries all available routing information and the incremental dump that only carries the information that has changed since the last dump.

Properties

Because DSDV is dependent on periodic broadcasts it needs some time to converge before a route can be used. This converge time can probably be considered negligible in a static wired network, where the topology is not changing so frequently. In an ad-hoc network on the other hand, where the topology is expected to be very dynamic, this converge time will probably mean a lot of dropped packets before a valid route is detected. The periodic broadcasts also add a large amount of overhead into the network.

2.3.4 Ad-hoc On Demand Distance vector - AODV

Description

The Ad Hoc On-Demand Distance Vector (AODV) (21) routing protocol enables multi-hop routing between participating mobile nodes wishing to establish and maintain an ad-hoc network. AODV is based upon the distance vector algorithm. The difference is that AODV is reactive, as opposed to proactive protocols like DV, i.e. AODV only requests a route when needed and does not require nodes to maintain routes to destinations that are not actively used in communications. As long as the endpoints of a communication connection have valid routes to each other, AODV does not play any role.

Features of this protocol include loop freedom and that link breakages cause immediate notifications to be sent to the affected set of nodes, but only that set. Additionally, AODV has support for multicast routing and avoids the Bellman Ford "counting to infinity" problem. The use of destination sequence numbers guarantees that a route is "fresh".

The algorithm uses different messages to discover and maintain links. Whenever a node wants to try and find a route to another node, it broadcasts a Route Request (RREQ) to all its neighbors. The RREQ propagates through the network until it reaches the destination or a node with a fresh enough route to the destination. Then the route is made available by unicasting a RREP back to the source.

The algorithm uses hello messages (a special RREP) that are broadcasted periodically to the immediate neighbors. These hello messages are local advertisements for the continued presence of the node and neighbors using routes through the broadcasting node will continue to mark the routes as valid. If hello messages stop coming from a particular node, the neighbor can assume that the node has moved away and mark that link to the node as broken and notify the affected set of nodes by sending a link failure notification (a special RREP) to that set of nodes.

Properties

The advantage with AODV compared to classical routing protocols like distance vector and link-state is that AODV has greatly reduced the number of routing messages in the network. AODV achieves this by using a reactive approach. This is probably necessary in an ad-hoc network to get reasonably performance when the topology is changing often.

AODV is also routing in the more traditional sense compared to for instance source routing based proposals like DSR. The advantage with a more traditional routing protocol in an ad-hoc network is that connections from the ad-hoc network to a wired network like the Internet is most likely easier.

The sequence numbers that AODV uses represents the freshness of a route and is increased when something happens in the surrounding area. The sequence prevents loops from being formed, but can however also be the cause for new problems. What happens for instance when the sequence numbers no longer are synchronized in the network? This can happen when the network becomes partitioned, or the sequence numbers wrap around.

AODV only support one route for each destination. It should however be fairly easy to modify AODV, so that it supports several routes per destination. Instead of requesting a new route when an old route becomes invalid, the next stored route to that destination could be tried. The probability for that route to still be valid should be rather high.

Although the Triggered Route Replies are reduced in number by only sending the Triggered Route Replies to affected senders, they need to traverse the whole way from the failure to the senders. This distance can be quite high in numbers of hops. AODV sends one Triggered RREP for every active neighbor in the active neighbor list for all entries that have been affected of a link failure. This can mean that each active neighbor can receive several triggered RREPs informing about the same link failure, but for different destinations, if a large fraction of the network traffic is routed through the same node and this node goes down. An aggregated solution would be more appropriate here.

AODV uses hello messages at the IP-level. This means that AODV does not need support from the link layer to work properly. It is however questionable if this kind of protocol can operate with good performance without support from the link layer. The hello messages adds a significant overhead to the protocol.

AODV does not support unidirectional links. When a node receives a RREQ, it will setup a reverse route to the source by using the node that forwarded the

RREQ as nexthop. This means that the route reply, in most cases is unicasted back the same way as the route request used. Unidirectional link support would make it possible to utilize all links and not only the bi-directional links. It is however questionable if unidirectional links are desirable in a real environment. The acknowledgements in the MAC protocol IEEE 802.11 would for instance not work with unidirectional links.

2.3.5 Dynamic Source Routing - DSR

Description

Dynamic Source Routing (DSR) (22) (14) (15) also belongs to the class of reactive protocols and allows nodes to dynamically discover a route across multiple network hops to any destination. Source routing means that each packet in its header carries the complete ordered list of nodes through which the packet must pass. DSR uses no periodic routing messages (e.g. no router advertisements), thereby reducing network bandwidth overhead, conserving battery power and avoiding large routing updates throughout the ad-hoc network. Instead DSR relies on support from the MAC layer (the MAC layer should inform the routing protocol about link failures).

Properties

DSR uses the key advantage of source routing. Intermediate nodes do not need to maintain up-to-date routing information in order to route the packets they forward. There is also no need for periodic routing advertisement messages, which will lead to reduce network bandwidth overhead, particularly during periods when little or no significant host movement is taking place. Battery power is also conserved on the mobile hosts, both by not sending the advertisements and by not needing to receive them, a host could go down to sleep instead.

This protocol has the advantage of learning routes by scanning for information in packets that are received. A route from A to C through B means that A learns the route to C, but also that it will learn the route to B. The source route will also mean that B learns the route to A and C and that C learns the route to A and B. This form of active learning is very good and reduces overhead in the network.

However, each packet carries a slight overhead containing the source route of the packet. This overhead grows when the packet has to go through more hops to reach the destination. So the packets sent will be slightly bigger, because of the overhead.

Running the interfaces in promiscuous mode is a serious security issue. Since the address filtering of the interface is turned off and all packets are scanned for information. A potential intruder could listen to all packets and scan them for useful information such as passwords and credit card numbers. Applications have to provide the security by encrypting their data packets before transmission. The routing protocols are prime targets for impersonation attacks and must therefore also be encrypted.

DSR also has support for unidirectional links by the use of piggybacking the source route a new request. This can increase the performance in scenarios where we have a lot of unidirectional links. We must however have a MAC protocol that also supports this.

2.3.6 Zone Routing Protocol - ZRP

Description

Zone Routing Protocol (ZRP) (13) is a hybrid of a reactive and a proactive protocol. It divides the network into several routing zones and specifies two totally detached protocols that operate inside and between therouting zones.

The Intrazone Routing Protocol (IARP) operates inside the routing zone and learns the minimum distance and routes to all the nodes within the zone. The protocol is not defined and can include any number of proactive protocols, such as Distance Vector or link-state routing. Different zones may operate with different intrazone protocols as long as the protocols are restricted to those zones. A change in topology means that update information only propagates within the affected routing zones as opposed to affecting the entire network.

The second protocol, the Interzone Routing Protocol (IERP) is reactive and is used for finding routes between different routing zones. This is useful if the destination node does not lie within the routing zone.

The protocol then broadcasts (i.e. bordercasts) a Route REQuest (RREQ) to all border nodes within the routing zone, which in turn forwards the request if the destination node is not found within their routing zone. This procedure is repeated until the requested node is found and a route reply is sent back to the source indicating the route. IERP uses a Bordercast Resolution Protocol (BRP) (13) that is included in ZRP. BRP provides bordercasting services, which do not exist in IP. Bordercasting is the process of sending IP datagrams from one node to all its peripheral nodes. BRP keeps track of the peripheral nodes and resolves a border cast address to the individual IP-addresses of the peripheral nodes. The message that was bordercasted is then encapsulated into a BRP packet and sent to each peripheral node.

Properties

ZRP is a very interesting protocol and can be adjusted of its operation to the current network operational conditions (e.g. change the routing zone diameter). However this is not done dynamically, but instead it is suggested that this zone radius should be set by the administration of the network or with a default value by the manufacturer. The performance of this protocol depends quite a lot on this decision.

Since this is a hybrid between proactive and reactive schemes, this protocol use advantages from both. Routes can be found very fast within the routing zone, while routes outside the zone can be found by efficiently querying selected nodes in the network. One problem is however that the proactive intrazone routing protocol is not specified. The use of different intrazone routing protocols would mean that the nodes would have to support several different routing protocols. This is not a good idea when dealing with thin clients. It is better to use the same intrazone routing protocol in the entire network.

ZRP also limits propagation of information about topological changes to the neighborhood of the change only (as opposed to a fully proactive scheme, which would basically flood the entire network when a change in topology occurred). However, a change in topology can affect several routing zones.

2.3.7 Temporally-Ordered Routing Algorithm - TORA

Description

Temporally Ordered Routing Algorithm (TORA) (23) (24) is a distributed routing protocol. The basic underlying algorithm is one in a family referred to as link reversal algorithms. TORA is designed to minimize reaction to topological changes. A key concept in its design is that control messages are typically localized to a very small set of nodes. It guarantees that all routes are loop-free (temporary loops may form), and typically provides multiple routes for any source/destination pair. It provides only the routing mechanism and depends on Internet MANET Encapsulation Protocol for other underlying functions.

TORA can be separated into three basic functions: creating routes, maintaining routes, and erasing routes. The creation of routes basically assigns directions to links in an undirected network or portion of the network, building a directed acyclic graph (DAG) rooted at the destination (See Figure 3).

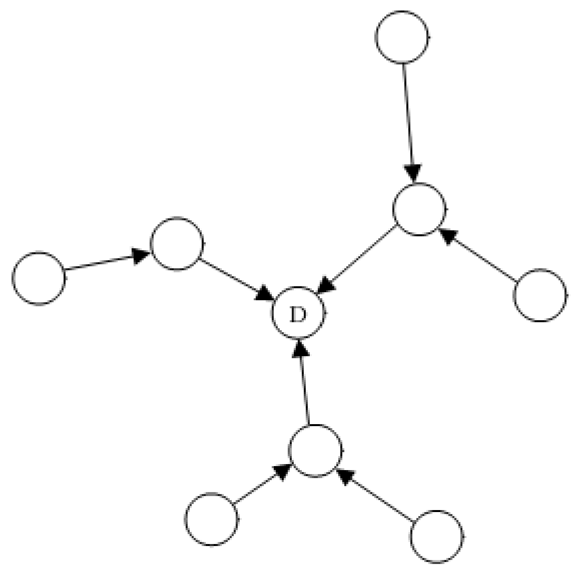


Figure 3 Directed acyclic graph rooted at destination.

TORA associates a height with each node in the network. All messages in the network flow downstream, from a node with higher height to a node with lower height. Routes are discovered using Query (QRY) and Update (UPD) packets. When a node with no downstream links needs a route to a destination, it will broadcast a QRY packet. This QRY packet will propagate through the network until it reaches a node that has a route or the destination itself. Such a node will then broadcast a UPD packet that contains the node height. Every node receiving this UPD packet will set its own height to a larger height than specified in the UPD message. The node will then broadcast its own UPD packet. This will result in a number of directed links from the originator of the QRY packet to the destination. This process can result in multiple routes.

Maintaining routes refers to reacting to topological changes in the network in a manner such that routes to the destination are re-established within a finite time, meaning that its directed portions return to a destination-oriented graph within a finite time. Upon detection of a network partition, all links in the portion of the network that has become partitioned from the destination are marked as undirected to erase invalid routes. The erasing of routes is done using clear (CLR) messages.

Properties

The protocols underlying link reversal algorithm will react to link changes through a simple localized single pass of the distributed algorithm. This prevents CLR packets to propagate too far in the network. A comparison made by the CMU Monarch project has however shown that the overhead in TORA is quite large because of the use of IMEP.

The graph is rooted at the destination, which has the lowest height. However, the source originating the QRY does not necessarily have the highest height. This can lead to the situation, where multiple routes are possible from the source to the destination, but only one route will be discovered. The reason for this is that the height is initially based on the distance in number of hops from the destination.

2.3.7 Optimized Link State Routing Protocol - OLSR

Description

The Optimized Link State Routing Protocol (OLSR) is developed for mobile ad hoc networks. It operates as a table driven, proactive protocol, i.e., exchanges topology information with other nodes of the network regularly. Each node selects a set of its neighbor nodes as "multipoint relays" (MPR). In OLSR, only nodes, selected as such MPRs, are responsible for forwarding control traffic, intended for diffusion into the entire network. MPRs provide an efficient mechanism for flooding control traffic by reducing the number of transmissions required.

Nodes, selected as MPRs, also have a special responsibility when declaring link state information in the network. Indeed, the only requirement for OLSR to provide shortest path routes to all destinations is that MPR nodes declare link-state information for their MPR selectors. Additional available link-state information may be utilized, e.g., for redundancy.

Nodes which have been selected as multipoint relays by some neighbor node(s) announce this information periodically in their control messages. Thereby a node announces to the network, that it has reachability to the nodes which have selected it as an MPR. In route calculation, the MPRs are used to form the route from a given node to any destination in the network. Furthermore, the protocol uses the MPRs to facilitate efficient flooding of control messages in the network.

A node selects MPRs from among its one hop neighbors with "symmetric", i.e., bi-directional, linkages. Therefore, selecting the route through MPRs automatically avoids the problems associated with data packet transfer over uni-directional links (such as the problem of not getting link-layer acknowledgments for data packets at each hop, for link-layers employing this technique for unicast traffic). OLSR is developed to work independently from other protocols. Likewise, OLSR makes no assumptions about the underlying link-layer.

MRP – Multipoint Relay

The idea of multipoint relays (25) is to minimize the overhead of flooding messages in the network by reducing redundant retransmissions in the same region. Each node in the network selects a set of nodes in its symmetric 1-hop neighborhood which may retransmit its messages. This set of selected neighbor nodes is called the "Multipoint Relay" (MPR) set of that node. The neighbors of node N which are not in its MPR set, receive and process broadcast messages but do not retransmit broadcast messages received from node N. Each node selects its MPR set from among its 1-hop symmetric neighbors. This set is selected such that it covers (in terms of radio range) all symmetric strict 2-hop nodes. The MPR set of N, denoted as MPR(N), is then an arbitrary subset of the symmetric 1-hop neighborhood of N which satisfies the following condition: every node in the symmetric strict 2-hop neighborhood of N must have a symmetric link towards MPR(N).

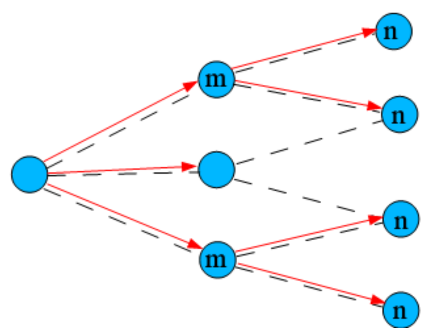


Figure 4 MRP Forwarding: Node m is the MRP set of source node

The smaller a MPR set, the less control traffic overhead results from the routing protocol. gives an analysis and example of MPR selection algorithms. Each node maintains information about the set of neighbors that have selected it as MPR. This set is called the "Multipoint Relay Selector set" (MPR selector set) of a node.

A node obtains this information from periodic HELLO messages received from the neighbors. A broadcast message, intended to be diffused in the whole network, coming from any of the MPR selectors of node N is assumed to be retransmitted by node N, if N has not received it yet. This set can change over time (i.e., when a node selects another MPR-set) and is indicated by the selector nodes in their HELLO messages.

MRP forwarding gives certain edge over ordinary flooding. That's where OLSR gains over some other flooding based protocols. Figure 5 shows an example comparison between ordinary flooding and MRP flooding.

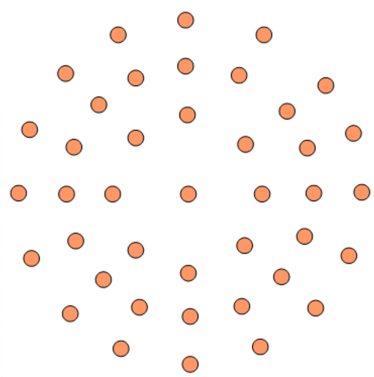


Figure 5.a Network topology

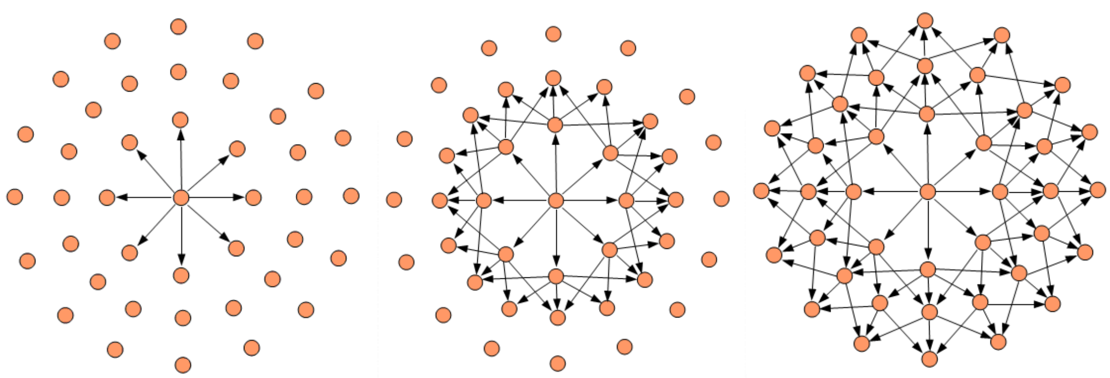


Figure 5.b Ordinary Flooding

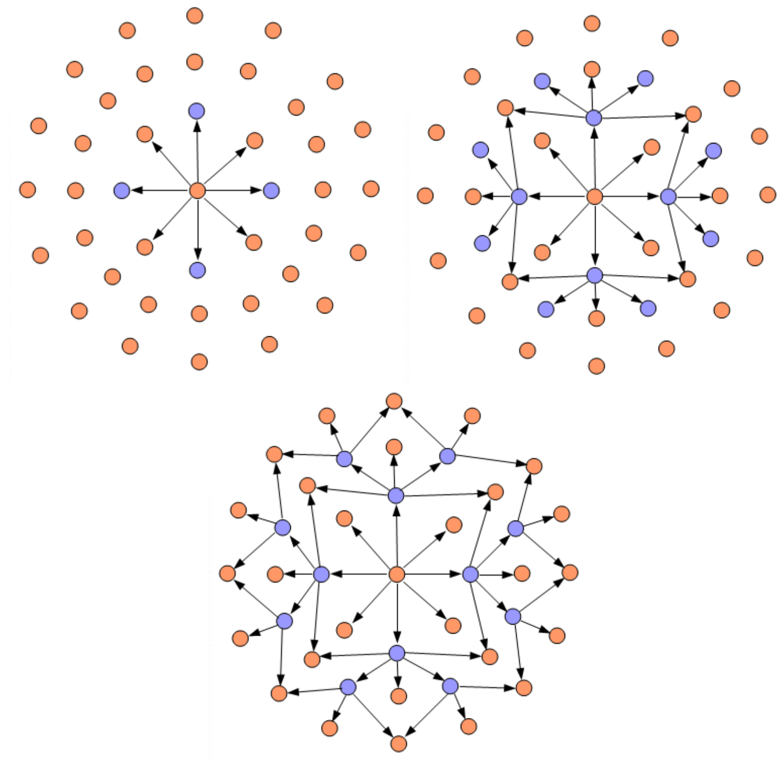


Figure 5.c MRP Flooding

Figure 5 Comparison between flooding and MRP

Applicability

OLSR is a proactive routing protocol for mobile ad-hoc networks (MANETs) (19) (26). It is well suited to large and dense mobile networks, as the optimization achieved using the MPRs works well in this context. The larger and more dense a network, the more optimization can be achieved as compared to the classic link state algorithm. OLSR uses hop-by-hop routing, i.e., each node uses its local information to route packets. OLSR is well suited for networks, where the traffic is random and sporadic between a larger set of nodes rather than being almost exclusively between a small specific set of nodes. As a proactive Clausen &

Jacquet Experimental [Page 7] RFC 3626 Optimized Link State Routing October 2003 protocol, OLSR is also suitable for scenarios where the communicating pairs change over time: no additional control traffic is generated in this situation since routes are maintained for all known destinations at all times.

2.4 SIP: Session Initiation Protocol

2.4.1 Overview

Session Initiation Protocol (SIP) (27) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543) that can be used to establish, maintain, and terminate calls between two or more end points.

Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. *Signaling* allows call information to be carried across network boundaries. *Session management* provides the ability to control the attributes of an end-to-end call.

SIP provides the capabilities to:

- Determine the location of the target end point—SIP supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target end point—Via Session Description Protocol (SDP), SIP determines the "lowest level" of common services between the end points. Conferences are established using only the media capabilities that can be supported by all end points.
- Determine the availability of the target end point—If a call cannot be completed because the target end point is unavailable, SIP determines whether the called party is already on the phone or did not answer in the

allotted number of rings. It then returns a message indicating why the target end point was unavailable.

- Establish a session between the originating and target end point—If the call can be completed, SIP establishes a session between the end points. SIP also supports mid-call changes, such as the addition of another end point to the conference or the changing of a media characteristic or codec.
- Handle the transfer and termination of calls—SIP supports the transfer of calls from one end point to another. During a call transfer, SIP simply establishes a session between the transferee and a new end point (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.

Conferences can consist of two or more users and can be established using multicast or multiple unicast sessions.

2.4.2 Components of SIP

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can function in one of the following roles:

- User agent client (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

Typically, a SIP end point is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiated the request.

From an architecture standpoint, the physical components of a SIP network can be grouped into two categories: clients and servers. Figure 6 illustrates the architecture of a SIP network.

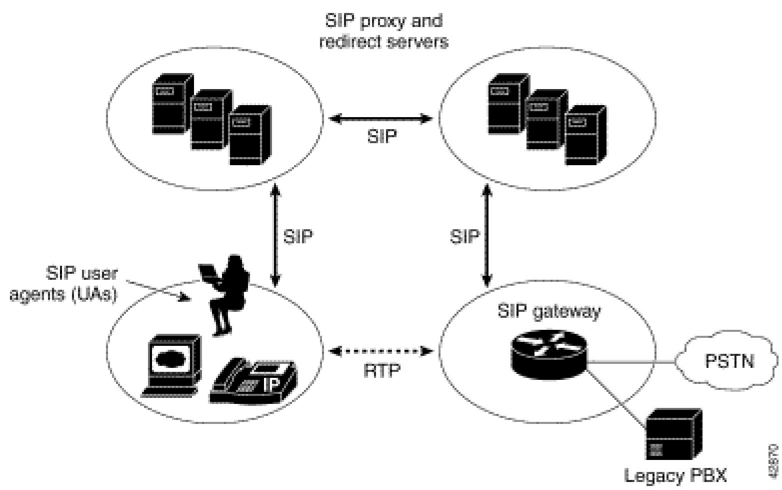


Figure 6 SIP Architecture

SIP Clients

SIP clients include:

- Phones—Can act as either a UAS or UAC. Softphones (PCs that have phone capabilities installed) and Cisco SIP IP phones can initiate SIP requests and respond to requests.

- Gateways—Provide call control. Gateways provide many services, the most common being a translation function between SIP conferencing endpoints and other terminal types. This function includes translation between transmission formats and between communications procedures. In addition, the gateway translates between audio and video codecs and performs call setup and clearing on both the LAN side and the switched-circuit network side.

SIP Servers

SIP servers include:

- Proxy server—The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.

- Redirect server—Provides the client with information about the next hop or hops that a message should take and then the client contacts the next hop server or UAS directly.

- Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.

2.4.3 How SIP Works

SIP is a simple, ASCII-based protocol that uses requests and responses to establish communication among the various components in the network and to ultimately establish a conference between two or more end points. (28)

Users in a SIP network are identified by unique SIP addresses. A SIP address is similar to an e-mail address and is in the format of sip:*userID@gateway*.com. The user ID can be either a user name or an E.164 address.

Users register with a registrar server using their assigned SIP addresses. The registrar server provides this information to the location server upon request.

When a user initiates a call, a SIP request is sent to a SIP server (either a proxy or a redirect server). The request includes the address of the caller (in the From header field) and the address of the intended callee (in the To header field). The following sections provide simple examples of successful, point-to-point calls established using a proxy and a redirect server.

Over time, a SIP end user might move between end systems. The location of the end user can be dynamically registered with the SIP server. The location server can use one or more protocols (including finger, rwhois, and LDAP) to locate the end user. Because the end user can be logged in at more than one station and because the location server can sometimes have inaccurate information, it might return more than one address for the end user. If the request is coming through a SIP proxy server, the proxy server will try each of the returned addresses until it locates the end user. If the request is coming through a SIP redirect server, the redirect server forwards all the addresses to the caller in the Contact header field of the invitation response.

Establishing A SIP Session Within the Same Domain

The diagram below (Figure 7) illustrates the establishment of a SIP session between two users who subscribe to the same IS and, hence, use the same domain. User A relies on a SIP phone. User B has a PC running a soft client that can support voice and video. Upon powering up, both users register their availability and their IP addresses with the SIP Proxy Server in the ISP's network. User A, who is initiating this call, tells the SIP Proxy Server he/she wan to contact User B. The SIP Proxy Server then asks for and receives User B's IP address from the SIP Registrar Server. The SIP Proxy Server relays User A's invitation to communicate with User B, including -- using SDP – the medium or media User A wants to use. User B informs the SIP Proxy Server that User A's invitation is acceptable and that he/she is ready to receive the message. The SIP Proxy Server communicates this to User A, establishing the SIP session. The users then create a point-to-point RTP connection enabling them to interact.

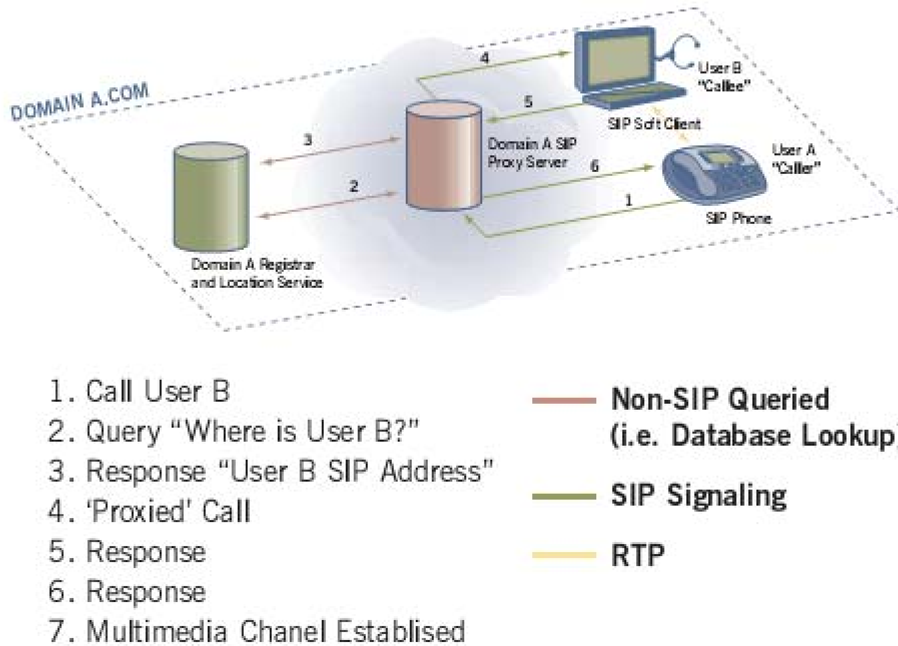
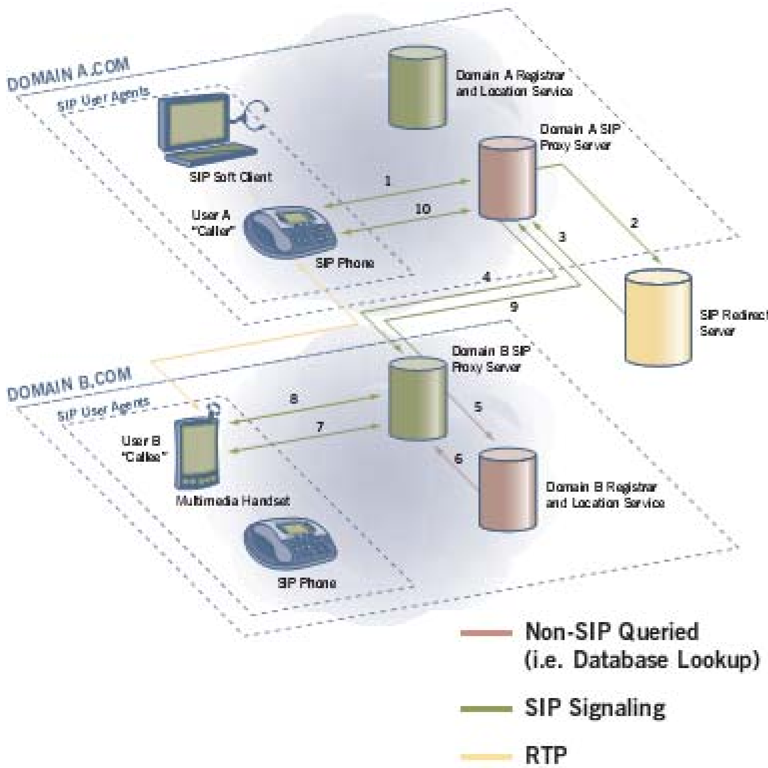


Figure 7 SIP Communication within same domain

Establishing A SIP Session In Dissimilar Domains

The difference between this scenario (Figure 8) and the first is that when User A invites User B -- who is now using a multimedia handset -- for a SIP session the SIP Proxy Server in Domain A recognizes that User B is outside its domain. The SIP Proxy Server then queries the SIP Redirect Server -- which can reside in either or both Domain A or B -- for User B's IP address. The SIP Redirect Server feeds User B's contact information back to the SIP Proxy Server, which forwards the SIP session invitation to the SIP Proxy Server in Domain B. The

Domain B SIP Proxy Server delivers User A's invitation to User B, who forwards his/her acceptance along the same path the invitation traveled.



1. Call User B
2. Query "How do I get to User B, Domain B?"
3. Response "Address of Proxy Controller for Domain"
4. Call 'Proxied' to SIP Proxy for Domain B
5. Query "Where is User B?"
6. User B's Address
7. Proxied Call
8. Response
9. Response
10. Response
11. Multimedia Channel Established

Figure 8 SIP Communication in different domain

2.5 Simulation Environment

For simulation purpose we have used NS2 (Network Simulator) (29) with an extension of OOLSR.

NS is an event driven network simulator developed at UC Berkeley that simulates variety of IP networks. It implements network protocols such as TCP and UPD, traffic source behavior such as FTP, Telnet, Web, CBR and VBR, router queue management mechanism such as Drop Tail, RED and CBQ, routing algorithms such as Dijkstra, and more. NS also implements multicasting and some of the MAC layer protocols for LAN simulations. The NS project is now a part of the VINT project that develops tools for simulation results display, analysis and converters that convert network topologies generated by well-known generators to NS formats. Currently, NS (version 2) written in C++ and OTcl (Tcl script language with Object-oriented extensions developed at MIT) is available.

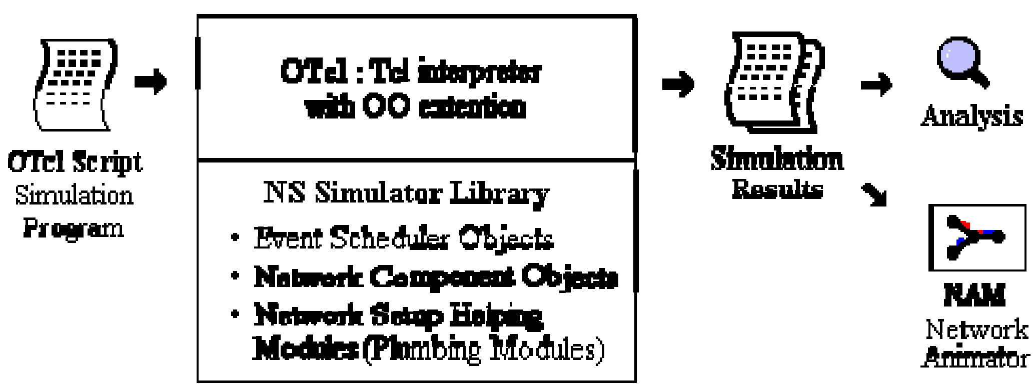


Figure 9 Simplified User's View of NS

As shown in Figure 9, in a simplified user's view, NS is Object-oriented Tcl (OTcl) script interpreter that has a simulation event scheduler and network component object libraries, and network setup (plumbing) module libraries (actually, plumbing modules are implemented as member functions of the base simulator object). In other words, to use NS, one has to program in OTcl script language. To setup and run a simulation network, a user should write an OTcl script that initiates an event scheduler, sets up the network topology using the network objects and the plumbing functions in the library, and tells traffic sources when to start and stop transmitting packets through the event scheduler. The term "plumbing" is used for a network setup, because setting up a network is plumbing possible data paths among network objects by setting the "neighbor" pointer of an object to the address of an appropriate object. When a user wants to make a new network object, he or she can easily make an object either by writing a new object or by making a compound object from the object library, and plumb the data path through the object. This may sound like complicated job, but the plumbing OTcl modules actually make the job very easy. The power of NS comes from this plumbing.

Another major component of NS beside network objects is the event scheduler. An event in NS is a packet ID that is unique for a packet with scheduled time and the pointer to an object that handles the event. In NS, an event scheduler keeps track of simulation time and fires all the events in the event queue scheduled for the current time by invoking appropriate network components, which usually are the ones who issued the events, and let them do the appropriate action associated with packet pointed by the event. Network components communicate with one another passing packets, however this does not consume actual simulation time. All the network components that need to spend some simulation time handling a packet (i.e. need a delay) use the event scheduler by issuing an event for the packet and waiting for the event to be fired to itself before doing further action handling the packet. For example, a network switch component that simulates a switch with 20 microseconds of switching delay issues an event for a packet to be switched to the scheduler as an event 20 microsecond later. The scheduler after 20 microsecond dequeues the event and fires it to the switch component, which then passes the packet to an appropriate output link component. Another use of an event scheduler is timer. For example, TCP needs a timer to keep track of a packet transmission time out for retransmission (transmission of a packet with the same TCP packet number but different NS packet ID). Timers use event schedulers in a similar manner that delay does. The only difference is that timer measures a time value associated with a packet and does an appropriate action related to that packet after a certain time goes by, and does not simulate a delay.

NS is written not only in OTcl but in C++ also. For efficiency reason, NS separates the data path implementation from control path implementations. In order to reduce packet and event processing time (not simulation time), the event scheduler and the basic network component objects in the data path are written and compiled using C++. These compiled objects are made available to the OTcl interpreter through an OTcl linkage that creates a matching OTcl object for each of the C++ objects and makes the control functions and the configurable variables specified by the C++ object act as member functions and member variables of the corresponding OTcl object. In this way, the controls of the C++ objects are given to OTcl. It is also possible to add member functions and variables to a C++ linked OTcl object. The objects in C++ that do not need to be controlled in a simulation or internally used by another object do not need to be linked to OTcl. Likewise, an object (not in the data path) can be entirely implemented in OTcl. Figure 10 shows an object hierarchy example in C++ and OTcl. One thing to note in the figure is that for C++ objects that have an OTcl linkage forming a hierarchy, there is a matching OTcl object hierarchy very similar to that of C++.

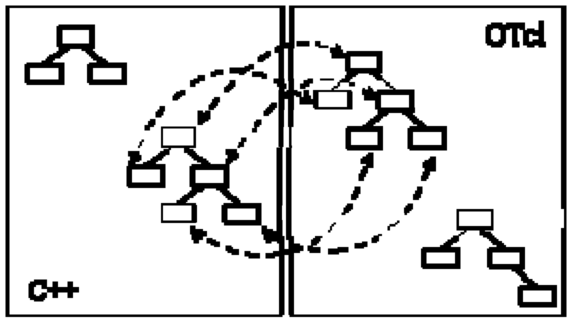


Figure 10 C++ and OTCL: The Duality

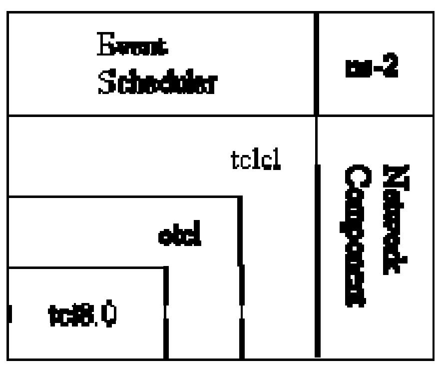


Figure 11 Architectural View of NS

Figure 11shows the general architecture of NS. In this figure a general user (not an NS developer) can be thought of standing at the left bottom corner, designing and running simulations in Tcl using the simulator objects in the OTcl library. The event schedulers and most of the network components are implemented in C++ and available to OTcl through an OTcl linkage that is implemented using tclcl. The whole thing together makes NS, which is an OO extended Tcl interpreter with network simulator libraries.

Chapter 3

Pseudo Cellular Architecture

3.1 Overview

Pseudo cellular architecture is a kind of emulation of GSM cellular architecture to facilitate SIP registration service and connectivity mechanism to ad-hoc environment. To support SIP registration service in ad-hoc environment for initiation of voice communication, works done so far either goes for total decentralized or total centralized mechanism. Our protocol lies in between. Every SIP enable node in our system is also capable of being a SIP registrar. We can generate two extreme possible scenarios. In one scenario all the nodes are registrar. In the other scenario, only a single node is the registrar. But an optimum (considering control overhead generated due to this protocol with respect to overall control overhead to maintain the routing protocol) solution may come when the whole topology is divided into some virtual "cells", within which a unique registrar server provides the registration service to all the nodes included in that cell. Generation of these registrar servers and allocation of nodes into cells is totally dynamic, topology independent and without the requirement of any special node.

3.2 Definition of Protocol Specific parameters

3.2.1 Cell

Cell is either a single node that has no neighbors or a maximal set of nodes those are currently registered to a common Registrar server. So cell grows around a registrar server with each new node registering onto that Registrar server. New cell is created with introduction of new Registrar server. In Figure 12 nodes are colored to show distinction between two different cells. Striped nodes are Registrar server of corresponding cell.

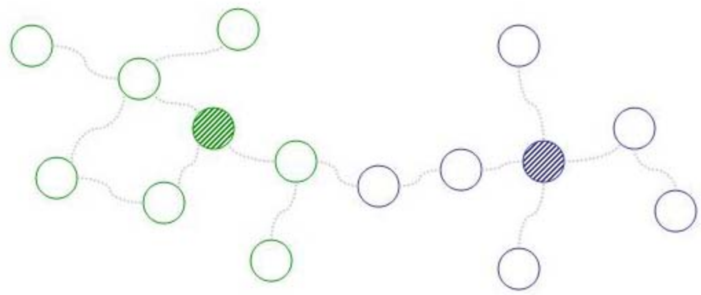


Figure 12 Example of Cell

3.2.2 Distance

Distance between two nodes under this protocol is the number of hops traversed from source node to destination node along the path that the routing algorithm provides. Two important aspects about this distance measure are:

- Distance according to this definition might not the shortest distance between the two nodes in the graph of the network. [i]

- Distances might not be equal in both directions.[ii]

Figure 13 shows distance between two neighboring registrar servers. Here the distance is 4.

[i] **Shortest Path Length** obviously is the lower limit of **Distance.**

[ii] Due to the directional nature of the wireless links, all routes are unidirectional and hence might differ when we switch source and destination. Different routes can easily have different distance value.

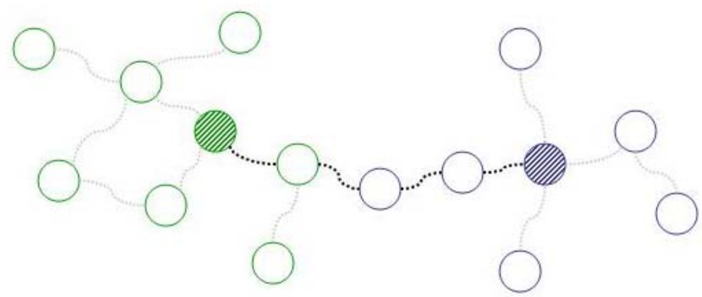


Figure 13 Example of Distance

3.2.3 Cell Diameter

The minimum amount of distance allowed between two neighboring Registrar servers denoted by $\boldsymbol{D_{cell}}$. It's a measure of the level of centralization. The greater the value of $\boldsymbol{D_{cell}}$, the lesser number of cells are generated in the topology, and the more centralized the service becomes.

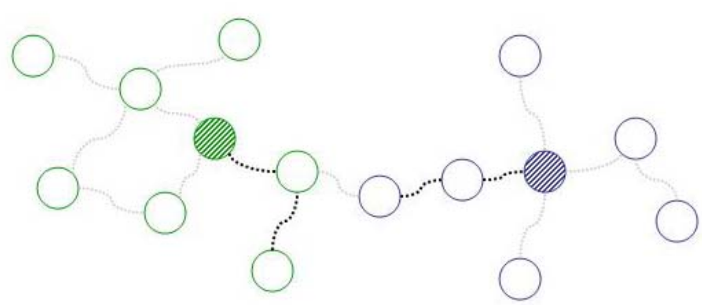


Figure 14 Illustration of cell diameter

3.2.4 Registrar Server Advertisement Interval

Time between two successive Registrar Server advertisement broadcasts, denoted by $\boldsymbol{T_{advertise}}$. It also controls the worst case waiting time before a newly joined node registers itself onto the network or declares itself as Registrar server. So increasing $\boldsymbol{T}_{advertise}$ decreases overall control overhead but causes poor system response time. On the contrary, decreasing $\boldsymbol{T}_{advertise}$, gives better network response with additional broadcast overhead.

3.2.5 Registration Timeout Interval

Each registration entry in the registrar server remains valid for a predefined time. This is registration timeout interval, denoted as $\boldsymbol{T}_{registration.}$ So even if a node registers itself to a server it must re-register itself after passing $\boldsymbol{T}_{registration}$ time. If we increase the value of this parameter, number of REGISTER messages generated is decreased but network response becomes worse[i].

3.2.6 Collision

Two registrar server is said to be in collision if they reside within a distance less than $\boldsymbol{D}_{cell}$.

3.2.7 Collision Set

A Collision Set is a maximal[ii] set of nodes where each node is a Registrar server and is in Collision with at least one other member of the same set.

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[i] When a registrar server goes down or loses its registration data somehow, all the nodes within that cell is barred calling others until registrar server is fed with valid data. So if we feed the server with **REIGSTER** with longer intervals, system down time also becomes larger.

[ii] By **maximal** we mean there is no other set of nodes that is both a collision set and a pure superset of this set.

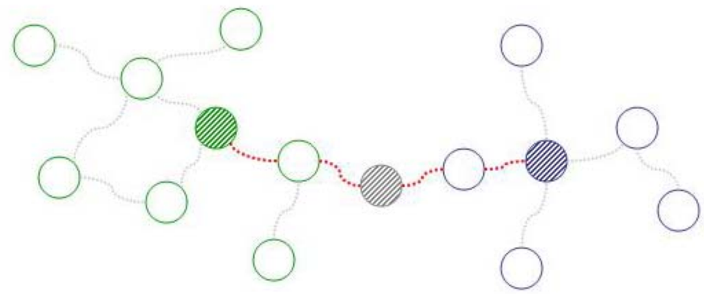


Figure 15 Collision Set of size 3

3.2.8 Intra-Cell Messages

REGISTER Unicast message. Being sent from a newly joined node to the Registrar Server of the cell. Contains *SIP URI* and *IP address* of the joining node.

QUERY Unicast message. Being sent from a SIP enabled node to Registrar Server. Contains destination *SIP URI*.

RESPONSE Unicast message. Sent from Registrar Server to the node asking QUERY. Contains *SIP URI* and *IP address* of the concerned node.

3.2.9 Inter-Cell Messages

ADERTISEMENT MPR Broadcast message. Sent from each Registrar Server periodically. Contains *IP Address* of the Registrar Server and a *measure of strength*[i] of the server.

QUERY[i] MPR Broadcast message. Sent from some Registrar servers. Contains destination *SIP URI*.

---

[i] When two registrars become active in a single cell, one must go down to subside ambiguity. So we need some sort of pre-agreed resolving scheme depending upon some measurable parameter of each server for this kind of situation. We call this *measure of strength* of a registrar server.

RESPONSE Unicast message. Sent from destination Cell registrar server to source Cell registrar server.

3.3 Protocol Details

3.3.1 Scenarios

Case 1: Node comes online, finds no neighbor. So it can safely declare itself as a registrar server. It then registers itself in its own registrar server.

Case 2: Node comes online and finds some neighbors. The daemon waits $T_{advertisement}$ amount of time to hear from some registrar server.

Case 2.1: Advertisement heard. So there are some registrar servers already on the network. Calculate distance (D_{id}) of each Registrar server from the advertisements heard.

Case 2.1.1: If $\min(D_{id}) < D_{cell}$, send REGISTER message to the registrar server with $\min(D_{id})$.

Case 2.1.2 If $\min(D_{id}) \geq D_{cell}$ start Registrar service in this node.

Case 2.2 No registrar server heard from, start Registrar service in this node.

Case 3 Node is already online.

Case3.1 Node was already a registrar server. Periodically (with $T_{advertisement}$ interval) broadcast ADVERTISEMENT message. Also listen to all

---

[i] Intra cell **QUERY** and **RESPONSE** is required because when we divide network into cells, registrar server of a cell only contains Registration information of that cell's nodes. So when one node of a cell tries to connect with a node of a different cell the registrar server of the source cell fails to resolve the SIP URI and is forced to "ask" other registrar servers on the network.

advertisement messages coming from other registrar servers of the network. If one or more such advertisement comes from a node with Distance less than Cell diameter, then we are presented with at least one cell that has a diameter less than our prescribed parameter. So we need to re arrange Registrar server assignment[i].

Case 3.2 Node was not a registrar server. So it must be included in some cell and registered to its Registrar server. Once registered to a Registrar server, a node stops looking for a better one (in terms of distance) and continually sends REGISTER message with a period $T_{registration}$. If the node doesn't receive ADVERTISEMENT from the registrar server within $T_{advertisement}$ time of last advertisement arrival, the node reverts back and follows Case 2.

3.3.2 Registrar Server Rearrangement

When a collision occurs, we can say distribution of Registrar servers is not distributed uniformly in the network. So we need to turn off few of them (and may be turn on few others) to make it well distributed. Like any other scenarios of the protocol, this is also done voluntarily with every node having equal priority.

When a registrar server detects a collision it tries to build a table of collision set sorted by considering *measure of strength* of the servers. On ideal case, if we have such table we can safely say that only the fittest one should survive and others should turn themselves off. But each registrar can only detect the subset of the collision set which only includes its neighbors. So if we apply the same procedure

---

[i] **Registrar server rearrangement** is discussed in detail later.

onto the subset of collision table on each colliding servers. It takes several iterations for the topology to settle down.

Figure 16 shows a common case of collision generation. Figure 17 and Figure 18 shows two cases of registrar server reassignment.

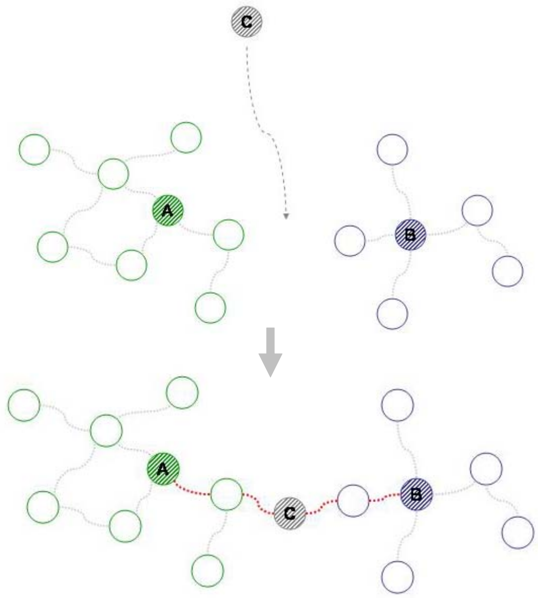
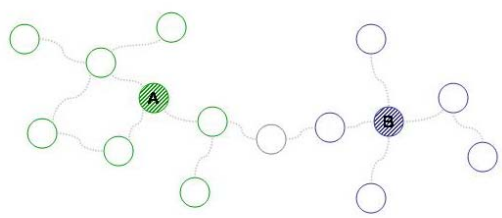
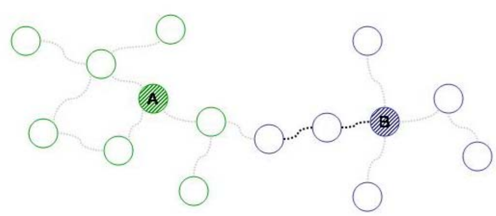


Figure 16 Creation of Collision: Registrar Server **C** enters the network due to mobility.

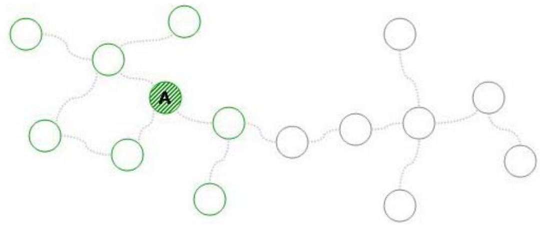


a) Server C goes down voluntarily

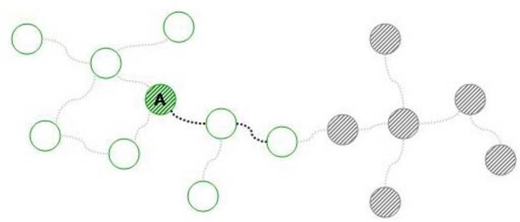


b) New node registers itself to Server B

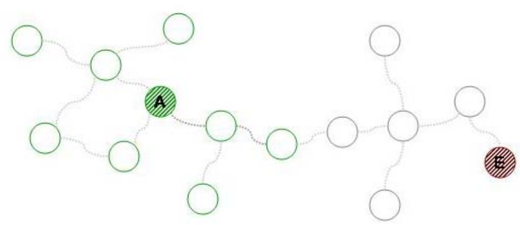
Figure 17 Rearrangement when A>B>C in terms of strength



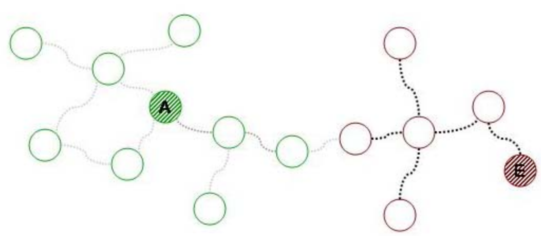
a) C goes down because A>C and B goes down because C>B



b) All the nodes outside $\boldsymbol{D_{cell}}$ decides to become server of the new Cell. Which creates new collision set



c) E is the strongest one of the new set so all others goes down.



d) Free nodes now register onto the new cell.

Figure 18 Rearrangement when A>C>B

3.4 State Diagram

According to this protocol each SIP enabled ad-hoc node can reside in four possible states. Transition between states are voluntary, deterministic and triggered upon receive of messages or timeouts.

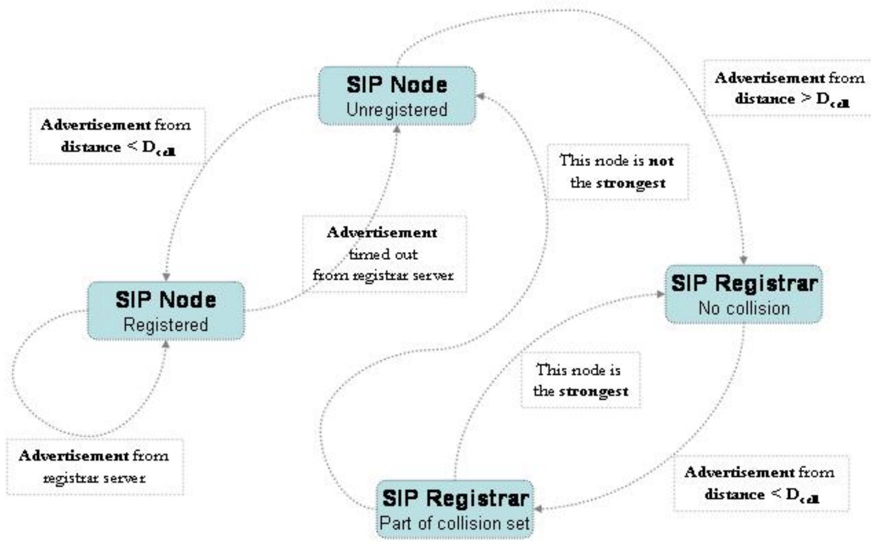


Figure 19 States of Nodes in this protocol

This chapter provided details on the protocol developed. On the next one we focus our attention on evaluating the performance of the protocol.

Chapter 4

Performance Evaluation

4.1 Performance Metric

Here we are trying to transmit voice data over some network. Quality of voice depends upon a lot of factors, and performance evaluation can be multidimensional. In our case we are trying to build some protocol that creates a SIP based architecture onto ad-hoc network, which eventually will support voice transmission. The design of SIP architecture doesn't have any direct influence over voice quality but since such protocol has to be maintained through some packet transfer between nodes, the more overhead it causes, the lesser the bandwidth remains for the user to use it for any actual purpose. So, our measurement of performance would be how much control overhead is required to actually maintain the protocol up and running.

$$\textit{Overhead ratio} = \frac{O_P}{O_T}$$

Where,

O_P = Control overhead due to this protocol

O_T = Control overhead generated due to the routing protocol

4.2 Qualitative Evaluation

Here we try to answer the question, why this protocol should perform better than the existing ones available? This protocol generates registrar server dynamically and divides the whole topology into cells. As such number of broadcasting nodes decreases and as a whole total overhead decreases. So in terms of control overhead this should perform better than total decentralized broadcast mechanism. On the other hand if we let a single registrar server handle the whole thing congestion will arise. So we need some load balance between multiple registrar servers. Dividing the topology into multiple cells helps this cause but introduces some additional overhead. This protocol provides for some parameters to control number and size of cells. By manipulating these numbers we can find the optimal balance between broadcast and congestion.

4.3 Quantitative Evaluation

We tried to evaluate the performance of the protocol by simulating some real world environment. For that first we need to set up the simulation environment.

4.3.1 Installation

To simulate and analyze performance of this protocol, following tools are to be installed.

- OLSR Implementation for NS2 (OOLSR-0.99.15)[i]
- Network Simulator (NS-2.27)[ii]
- Red Hat Linux 9.0[iii]

Following the standard procedure on favorable environment should be enough to install these tools.

4.3.2 Simulation Architecture

We need to simulate activity of a SIP registrar that is created dynamically from an ordinary SIP Node (able to run SIP User Agents). In this simulation registrar server is implemented as an application, which holds all the logic required for

---

[i] An object oriented implementation of OLSR protocol, available as a plugin of NS2. OOLSR is available a complete package including NS-allinone sources from (26)

[ii] OOLSR was developed for NS-2.27. No other distribution was tried on.

[iii] NS-2.27 ran successfully only on this distribution. Other distribution tried on was Fedora Core 3,5,6, Ubuntu Dapper Drake LTS. In case of Red Hat 9, the full package compiled only in KDE environment. Gnome seemed to have some problem with configuration.

running the protocol. This application requires some special packet structures to be sent in some special way that is not supported by ordinary UDP agents provided by NS2. So, an agent had to be built extending UDP. Which was eventually named sipUDPAgent. This agent utilizes underlying OLSR routing protocol to transmit packet to remote destinations.

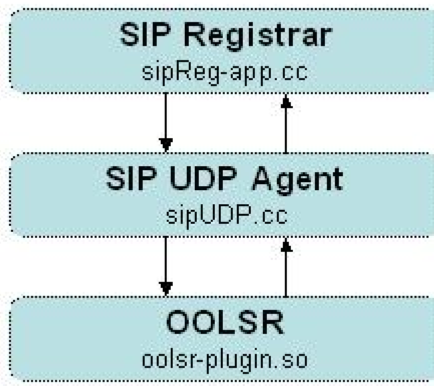


Figure 20 Simulation Architecture of the protocol

The simulation environment consists of wireless nodes randomly scattered and having random motion each with OLSR as routing protocol and SIP Registrar Application running. We let the setup run for a while and then parse the output trace file to find out how many packets were generated by our application and by the routing protocol. Performance can then be easily calculated. Similar simulation is done with different topologies and different protocol parameters to find out how well the protocol actually performs.

4.3.3 Component Details

At the time of report writing the simulation coding was still going on. What is presented here is still under construction and will go through a lot of modification before producing some result. So rather than focusing on overall functionality this section gives some insight on problems encountered during implementation and solves of the simulation system.

Registrar Server

SIPRegistrarApp.h

```
/*
 Author:    Nafees Ahmed
 File:      SIPRegistrarApp.h
 Written:   31/07/07 (for ns-2.27)
*/

#include "packet.h"
#include "app.h"
#include "sipUDP.h"
#include "timer-handler.h"
#include "hash_table.h"
#include <vector>
#include "node.h"


class TimeOutTimer;
class SIPRegistrarApp;

/*
 SIP Registration Information Datastructure
 timeout is provided for each individual record.
*/

typedef struct SIPregInfoType
{
 string  uri;
 nsaddr_t nodeAddr;
 TimeOutTimer*timer;
 bool  valid;
 int  hopCount;
} SIPRegInfo;


/*
 Advertisement timer.
 Extends TimerHandler that can schedule periodic events.
 On expiration it broadcasts ADVERTISEMENT message.
*/

class AdvTimer : public TimerHandler
{
 public:
 AdvTimer():TimerHandler(){}
 AdvTimer(SIPRegistrarApp* app_) : TimerHandler(), app(app_) {}
 inline virtual void expire(Event*);

 protected:
 SIPRegistrarApp* app;
};

/*
 Probe timer.
 Extends TimerHandler that can schedule periodic events.
 On expiration it checks for neighbor on interfaces available.
*/
class ProbeTimer : public TimerHandler
```

```
{
 public:
 ProbeTimer():TimerHandler(){}
 ProbeTimer(SIPRegistrarApp* app_) : TimerHandler(), app(app_) {}
 inline virtual void expire(Event*);

 protected:
 SIPRegistrarApp* app;
};

/*
 Timeout timer.
 Extends TimerHandler that can schedule periodic events.
 On expiration it delets the corresponding registration entry from
registration table.
*/
class TimeOutTimer : public TimerHandler
{
 public:
 TimeOutTimer():TimerHandler(){}
 TimeOutTimer(SIPRegInfo* info_) : TimerHandler(), info(info_) {}
 inline virtual void expire(Event*);
 protected:
 SIPRegInfo* info;
};

// SIP Registration Application
class SIPRegistrarApp : public Application
{
 public:
 SIPRegistrarApp();
 void advertise();      // called by AdvertiseTimer::expire
 void updateNeighborCount();  // called by NeighborProbeTimer::expire
 protected:
 int command(int argc, const char*const* argv);
 void start();          // Start application
 void stop();           // Stop application
 private:
 void init();
 virtual void recv_msg(int nbytes, const char *msg = 0); // (Sender/Receiver)

 int running_;              // If 1 advertisement is allowed
 double adv_int_;     // advertisement interval
 double neighbor_chk_int_; // probe for neighbor count check interval
 double item_timeout_int_; // period for registration information validity
 int neighbor_count;

 AdvTimer adv_timer_;
 ProbeTimer probe_timer_;
 int minDistance_;    // Protocol parameter : Cell diameter

 vector<SIPRegInfo> registrarList; // List of all registrar servers heard
from
 CHashTable<SIPRegInfo> regInfo; // Registration information table

 char SIPURI[20];     // SIP URI of this node
};
```

sipRegistrarApp.cc

```
/*
 Author:    Nafees Ahmed
 File:      SIPRegistrarApp.cc
 Written:   31/07/07 (for ns-2.27)
*/

#include "sipReg-app.h"

// When advertisement expires call advertisement sending function
void AdvTimer::expire(Event*)
{
 app->advertise();
}

// When probe timer expires probe for neighbor information
void ProbeTimer::expire(Event*)
{
 app->updateNeighborCount();
}

// When information times out, make it invalid
void TimeOutTimer::expire(Event*)
{
 info->valid = false;
}
// SIPRegistrarApp OTcl linkage class
static class SIPRegistrarAppClass : public TclClass
{
 public:
  SIPRegistrarAppClass() : TclClass("Application/VoWiFi/SIPRegistrar") {}
  TclObject* create(int, const char*const*) {
    return (new SIPRegistrarApp);
  }
} class_app_sip;



//small utility to color a NAM node. Used to identify registrar servers on
NAM output.Still on test.
void changeNodeColor(double atTime,const char * newColor,int id_)
{
 Tcl& tcl = Tcl::instance();
 int INTERVAL_COLOR_TIME = 5;
 int INTERVAL_COLOR = 1;

 if(atTime > 0.0 )
  atTime += 0.0000000001;

 atTime +=0.1;

// int i;
// for(i=0;i<INTERVAL_COLOR_TIME;i++)
// {
//  tcl.evalf("[Simulator instance] at %.9f {[Simulator instance] puts-nam-
traceall {n -t %.9f -s %d -S COLOR -c %s -o
%s}}",atTime,atTime,id_,"white","white");
//  atTime += INTERVAL_COLOR;
  tcl.evalf("[Simulator instance] at %.9f {[Simulator instance] puts-nam-
traceall {n -t %.9f -s %d -S COLOR -c %s -o
%s}}",atTime,atTime,id_,newColor,newColor);
```

```
//  atTime=atTime+INTERVAL_COLOR*INTERVAL_COLOR_TIME;
//}
}

/*
 Constructor
 Initializes instances of timers
 And binds OTcl variables to C++ variables.
*/
SIPRegistrarApp::SIPRegistrarApp() :
running_(0),adv_timer_(this),probe_timer_(this)
{
  bind("neighbor_chk_int", &neighbor_chk_int_);
  bind("adv_int", &adv_int_);
  bind("item_timeout_int",&item_timeout_int_);
  bind("minDistance",&minDistance_);
  strcpy(SIPURI,"MOBILE");
  bind("SIPURI",$SIPURI);
  bind("running", &running_);
}


// OTcl command interpreter
int SIPRegistrarApp::command(int argc, const char*const* argv)
{
  Tcl& tcl = Tcl::instance();

  if (argc == 3) {
    if (strcmp(argv[1], "attach-agent") == 0) {
      agent_ = (Agent*) TclObject::lookup(argv[2]);
      if (agent_ == 0) {
 tcl.resultf("no such agent %s", argv[2]);
 return(TCL_ERROR);
      }

      // Make sure the underlying agent support MM
      if(agent_->supportSIP())
      {
    agent_->enableSIP();
      }
      else
    {
   tcl.resultf("agent \"%s\" does not support SIP Application", argv[2]);
   return(TCL_ERROR);
      }

      agent_->attachApp(this);
      return(TCL_OK);
    }
  }
  return (Application::command(argc, argv));
}


/*
 initialize the application
 For now the URI is set to Node identifier of the node.
*/

void SIPRegistrarApp::init()
{
  char buf[30];
  sprintf(buf,"%d",agent_->addr());
```

```
  strcat(SIPURI,buf);
}

//starts the application
void SIPRegistrarApp::start()
{
  init();
  running_ = 0;

  probe_timer_.resched(adv_int_); //start probe timer
  adv_timer_.resched(adv_int_); //start advertisement timer

  Tcl& tcl = Tcl::instance();
  tcl.evalf("puts \"\ndevice %s online\"", SIPURI);
}

//stop the application
void SIPRegistrarApp::stop()
{
  running_ = 0;
}

/*
 Funtion for counting neighbor. Also holds part of the protocol logic.
 The neighbor count portion still doesn't work fully.
*/
void SIPRegistrarApp::updateNeighborCount()
{
 //do something to update neighbor count
 int neighborCount=0;
 Node* thisNode = Node::get_node_by_address(agent_->addr());
 neighbor_list_node* neighbor = thisNode->neighbor_list_;

 while(neighbor)
 {
  neighbor = neighbor->next;
  neighborCount++;
 }

 //check for proxy list and decide whether advertise for proxy service
 Tcl& tcl = Tcl::instance();
 tcl.evalf("puts \"\n\nnode %d \"",agent_->addr(),neighborCount);

 int minHopCount=99999;

 for(int i=0;i<registrarList.size();i++)
 {
   //reschedule the timer
   SIPRegInfo* info;
   info = &registrarList[i];
   tcl.evalf("puts \"\nentry %s :\"",info->uri.data());
   if(info!=0)
   {
    if(info->valid == true)
    {
     if(info->hopCount <= minHopCount )
     {
      minHopCount = info->hopCount;
     }
    }
   }
 }
```

```
 if(minHopCount > minDistance_)  //yes there is need for a new proxy server
 {
  running_ = true;
  changeNodeColor(Scheduler::instance().clock(),"red",agent_->addr());
 }
 else            //no need
 {
  running_ = false;
  changeNodeColor(Scheduler::instance().clock(),"black",agent_->addr());
 }

 probe_timer_.resched(neighbor_chk_int_);
}

/*
 Receive message from underlying agent
 Place for writing the whole protocol logic.
 Partially completed.
*/
void SIPRegistrarApp::recv_msg(int nbytes, const char *msg)
{
  int a;
  if(msg)
  {
    hdr_mm* mh_buf = (hdr_mm*) msg;
    if(agent_->addr()==0)
  a=0;
    switch (mh_buf->msgType)
    {
 case 1:  //advertisement broadcast received
 {
  bool found = false;
  if(strcmp("data-packet",mh_buf->SIPuri)==0)
   found = true;


  if(!found)
  {
  for(int i=0;i<registrarList.size();i++)
  {
   if(strcmp(registrarList[i].uri.data(),mh_buf->SIPuri)==0) //already there
is some entry
   {
    //reschedule the timer
    SIPRegInfo* info;
    info = &registrarList[i];
    if(info!=0)
    {
     info->valid = true;
     info->timer->resched(item_timeout_int_);
     info->hopCount = mh_buf->hopCount;
    }
    found = true;
   }
  }
  }
  if(!found) //if new entry is to be made
  {
   SIPRegInfo*    proxyInfo;
   proxyInfo  =  new SIPRegInfo();
   proxyInfo->uri  =mh_buf->SIPuri;
   proxyInfo->nodeAddr=mh_buf->nodeAddr;
   proxyInfo->valid =true;
```

```
    proxyInfo->hopCount = mh_buf->hopCount;
    TimeOutTimer* t = new TimeOutTimer(proxyInfo);
    t->resched(item_timeout_int_);
    proxyInfo->timer = t;
    registrarList.push_back(*proxyInfo);
   }

   //resolveProxyAssignment();
  }
  break;

  case 2:  //register

  break;

  case 3:

  break;
     }
   }
}

//ADVERTISEMENT broadcast function
void SIPRegistrarApp::advertise(void)
{
  double local_time = Scheduler::instance().clock();
  Tcl& tcl = Tcl::instance();

  int a;
  if(agent_->addr()==49)
   a=0;
  if(running_)
  {
 hdr_mm adv_buf;
 adv_buf.msgType =1;
 adv_buf.nodeAddr = agent_->addr();
 strcpy(adv_buf.SIPuri,SIPURI);
 adv_buf.nbytes = 40;

 // send advertise message
 //agent_->sendmsg(adv_buf.nbytes, (char*) &adv_buf);
 ((UdpMmAgent*)agent_)->sendToAll(adv_buf.nbytes,(char*) &adv_buf);
 //tcl.evalf("\nputs \"Message from %d, sent to %d\"",frmAddress,toAddress);
 //changeNodeColor(Scheduler::instance().clock(),"red",agent_->addr());
  }
  else
  {
   //changeNodeColor(Scheduler::instance().clock(),"black",agent_->addr());
  }
  //schedule next ACK time
  adv_timer_.resched(adv_int_);
}
```

UDP Agent

sipUDP.h

```
/*
 Author:    Nafees Ahmed
 File:      sipUDP.h
 Written:   31/07/07 (for ns-2.27)
*/



#include "udp.h"
#include "ip.h"
#include <string>
#include "config.h"

#define MAXTTL 65
// Header Structure
struct hdr_mm
{
 int   msgType;
 nsaddr_t nodeAddr; //source node address
 char  SIPuri[30]; //SIP uri
 int   hopCount; //# of hops traveled

 // Packet header access functions
        static int offset_;
        inline static int& offset() {return offset_;}
        inline static hdr_mm* access(const Packet* p){
                return (hdr_mm*) p->access(offset_);
        }
};


// sipUDPAgent Class definition
class sipUDPAgent : public UdpAgent {
public:
 UdpMmAgent();
 UdpMmAgent(packet_t);
 virtual int supportSIP() { return 1; }
 virtual void enableSIP() { support_mm_ = 1; }
 virtual void sendmsg(int nbytes, const char *flags = 0);
 void sendTo(int nbytes,const char *msg,nsaddr_t node);
 void sendToAll(int nbytes,const char *msg);
 void recv(Packet*, Handler*);
 void setPort(nsaddr_t);
protected:
 int support_mm_; // set to 1 if above is sipAPP
};

#endif
```

sipUDP.cc

```
/*
 Author:    Nafees Ahmed
 File:      sipUDP.cc
 Written:   31/07/07 (for ns-2.27)
*/

#include "sipUDP.h"
#include "rtp.h"
#include "random.h"
#include <string.h>


int hdr_mm::offset_;

// SIP Header Class
static class SIPHeaderClass : public PacketHeaderClass {
public:
 SIPHeaderClass() : PacketHeaderClass("PacketHeader/SIP",
           sizeof(hdr_mm)) {
  bind_offset(&hdr_mm::offset_);
 }
} class_mmhdr;


// sipUDPAgent OTcl linkage class
static class sipUDPAgentClass : public TclClass {
public:
 sipUDPAgentClass() : TclClass("Agent/UDP/sipUDP") {}
 TclObject* create(int, const char*const*) {
  return (new sipUDPAgent());
 }
} class_udpmm_agent;


// Constructor (with no arg)
sipUDPAgent::sipUDPAgent() : UdpAgent()
{
 support_mm_ = 0;

}

sipUDPAgent::sipUDPAgent(packet_t type) : UdpAgent(type)
{
 support_mm_ = 0;

}

/*
 Function so send UDP packet to a specific destination
 Destination address is of type nsaddr_t which is basically alias of int.
*/

void sipUDPAgent::sendTo(int nbytes, const char* msg,nsaddr_t dst)
{
 Tcl& tcl = Tcl::instance();
 nsaddr_t frmAddress= this->addr();
 nsaddr_t frmPort = this->port();
 this->dst_.addr_ = dst;
 nsaddr_t toAddress= this->daddr();
 nsaddr_t toPort= this->dport();
```

```
 //tcl.evalf("\nputs \"Message from node %d, port %d , to node %d , port
%d\"",frmAddress,frmPort,toAddress,toPort);

// Packet *p;
// p = allocpkt();
// target_->recv(p);
 sendmsg(nbytes,msg);
 idle();
}

/*
 function to implement MPR flooding.
 This function will only work if underlying routing protocol provides for
 MPR. At the time of code writing OOLSR didn't implement MPR or anyother
 broadcast mechanism.
 So it OOLSR had to be modified to support MPR.
*/

void sipUDPAgent::sendToAll(int nbytes, const char* msg)
{
 /*
 for(int i=0;i<50;i++)
 {
  if(this->addr()==i)
   continue;
  sendTo(nbytes,msg,i);
  idle();
 }
 */
 //hdr_ip* ih = (hdr_ip*)(msg);
 this->dst_.addr_ = IP_BROADCAST;
 Tcl& tcl = Tcl::instance();
 tcl.evalf("\nputs \"broadcast  from %d\"",this->addr());
 sendmsg(nbytes,msg);
}

// Method for sending message
void sipUDPAgent::sendmsg(int nbytes, const char* flags)
{
 Tcl& tcl = Tcl::instance();
 Packet *p;
 if (nbytes == -1) {
  printf("Error:  sendmsg() for UDPmm should not be -1\n");
  return;
 }
 double local_time = Scheduler::instance().clock();

  p = allocpkt();
  hdr_cmn::access(p)->size() = nbytes;
  //to eliminate recv to use MM fields for non MM packets
  hdr_mm* mh = hdr_mm::access(p);
  mh->ack = 0;
  mh->seq = 0;
  mh->nbytes = 0;
  mh->time = 0;
  mh->scale = 0;
  // mm udp packets are distinguished by setting the ip
  // priority bit to 15 (Max Priority).
  if(support_mm_)
  {
   hdr_ip* ih = hdr_ip::access(p);
   ih->prio_ = 15;
   ih->ttl_=MAXTTL;
```

```
    if(flags)
     memcpy(mh, flags, sizeof(hdr_mm));
   }

   target_->recv(p);
 idle();
}


// Receive funtion
void sipUDPAgent::recv(Packet* p, Handler*)
{
 hdr_ip* ih = hdr_ip::access(p);
 int bytes_to_deliver = hdr_cmn::access(p)->size();



   if(app_) {  // if SIP Application exists

    hdr_mm* mh = hdr_mm::access(p);

    // update hop count
    mh->hopCount = MAXTTL - ih->ttl_;
    hdr_mm mh_buf;
    memcpy(&mh_buf, mh, sizeof(hdr_mm));
    app_->recv_msg(mh_buf.nbytes, (char*) &mh_buf);

   Tcl& tcl = Tcl::instance();
   nsaddr_t frmAddress= this->addr();
   nsaddr_t frmPort = this->port();
   //tcl.evalf("\nputs \"Message from node %d, port %d , hop
%d\"",frmAddress,frmPort,MAXTTL - ih->ttl_);
   }
   Packet::free(p);
}
```

4.3.4 Plugging into NS2

To use our protocol in NS2 simulation we need following modifications in specified files. (30)

Common/Packet.h

```
...
enum packet_t {
         PT_TCP,
         PT_UDP,
         ......
         // insert new packet types here
         PT_TFRC,
         PT_TFRC_ACK,
          PT_SIP,
         PT_NTYPE // This MUST be the LAST one
};
...
...
class p_info {
public:
         p_info() {
                  name_[PT_TCP]= "tcp";
                  name_[PT_UDP]= "udp";
               ...........
                  name_[PT_TFRC]= "tcpFriend";
                  name_[PT_TFRC_ACK]= "tcpFriendCtl";

                  name_[PT_SIP]="SIPRegistrar";
                  name_[PT_NTYPE]= "undefined";
         }
        .....
 }
```

Tcl/lib/ns-default.tcl

```
...
# defaults for SIPRegistrarApp
Application/VoWiFi/SIPRegistrar  set adv_int   0.5
Application/VoWiFi/SIPRegistrar  set neighbor_chk_int 0.1
Application/VoWiFi/SIPRegistrar  set item_timeout_int 1.0
Application/VoWiFi/SIPRegistrar  set minDistance 5
```

Makefile

```
 ...
 OBJ_CC = \
          tools/random.o tools/rng.o tools/ranvar.o common/misc.o
          common/timer-handler.o \
          common/scheduler.o common/object.o common/packet.o \
          ...
          VoWiFi/sipReg-app.o \
```

```
VoWiFi/sipUDP.o \
$(OBJ_STL)
```

4.3.5 Simulation

To run simulation on the protocol a basic topology was created.

```
#---------------------------------------------------------------------------
# Random.tcl
# Simulation script for SIP on OLSR
#---------------------------------------------------------------------------

#---------------------------------------------------------------------------
# Initialization
#---------------------------------------------------------------------------

# (possibly) Remove and create result directory
set dirName "test-unicast-result"
exec sh -c "rm -rf $dirName && mkdir $dirName"

# Default node configuration
set nodeConfig "no-log 0; log-none ; log-route 1"

# Load the OOLSR as plugin
load-plugin ./oolsr-plugin --output $dirName/ns2agent.log \
              multicast route packet-drop

#---------------------------------------------------------------------------
# Create a simulation, with wireless support. This is basic (see ns2 doc)
#---------------------------------------------------------------------------
set ns [new Simulator]

set val(chan) Channel/WirelessChannel
set val(prop) Propagation/TwoRayGround
set val(netif) Phy/WirelessPhy
set val(mac) Mac/802_11
set val(ifq) Queue/DropTail/PriQueue
set val(ll) LL
set val(ant) Antenna/OmniAntenna
set val(ifqlen) 50 ;#
set val(nn)     50 ;# nb mobiles
set val(rp) PLUGINPROTOCOL
set val(x) [expr $val(nn) *100.0 + 100.0]
set val(y) 1000

set topo [new Topography]
$topo load_flatgrid $val(x) $val(y)
set god [create-god $val(nn)]

#$ns use-newtrace
set tracefd [open $dirName/unicast.tr w]
$ns trace-all $tracefd

set namtrace [open $dirName/unicast.nam w]
$ns namtrace-all-wireless $namtrace $val(x) $val(y)

$ns node-config -adhocRouting $val(rp) \
    -llType $val(ll) \
    -macType $val(mac) \
```

```
    -ifqType $val(ifq) \
    -ifqLen $val(ifqlen) \
    -antType $val(ant) \
    -propType $val(prop) \
    -phyType $val(netif) \
    -channel [new $val(chan)] \
    -topoInstance $topo \
    -agentTrace ON \
    -routerTrace ON \
    -macTrace OFF \
    -movementTrace OFF

#---------------------------------------------------------------------------
# Create nodes with OOLSR agent
#---------------------------------------------------------------------------

for {set i 0} {$i < $val(nn)} {incr i} {
    set node($i) [$ns node]

    set agent($i) [new Agent/UDP/sipUDP]
    set app($i) [new Application/VoWiFi/SIPRegistrar]
    $ns attach-agent $node($i) $agent($i)
    $app($i) attach-agent $agent($i)

    $node($i) random-motion 1
    $node($i) set X_ [expr $i * 100.0]
    $node($i) set Y_ [expr 500.0 + ((($i * 93) % 21) - 10 ) * 10.0]
    $node($i) set Z_ 0.0
    $ns initial_node_pos $node($i) 20

#    [$node($i) set ragent_] set-config \
#       "$nodeConfig ; log-file-name $dirName/oolsr-node-$i.log"

}
for {set i 0} {$i < $val(nn)} {incr i} {
$ns connect $agent($i) $agent([expr $val(nn) - $i -1])
$ns at ($i) "$app($i) start"
}

#---------------------------------------------------------------------------
# Sending traffic
#---------------------------------------------------------------------------

#attach and start some CBR application to simulate phone calls.
#should be filled up at the time of simulation.
#modify  this  section  to  evaluate  performance  of  the  protocol  in  different
#situations.

#---------------------------------------------------------------------------
# Finishing procedure
#---------------------------------------------------------------------------

proc finishSimulation { } {
    global ns node val dirName

    # Log the final state of all the nodes
    for {set i 0} {$i < $val(nn)} {incr i} {
                      [$node($i) set ragent_] state "$dirName/oolsr-node-$i.final-state"
    }

    # Exit
    puts "Finished simulation."
    $ns halt
```

```
}

#-----------------------------------------------------------------------------
# Run the simulation
#-----------------------------------------------------------------------------

proc runSimulation {duration} {
    global ns finishSimulation
    $ns at $duration "finishSimulation"
    $ns run
}

# decide how long the simulation will run
runSimulation 500.0
```

This script is to be simulated by NS and then output trace files (.tr) and animator files (.nam) can be analyzed to verify the performance of the protocol. Since the protocol was not completed at the time of report writing, simulation result is unavailable.

Chapter 5

Conclusion

Ad-Hoc network is an emerging field in networking arena. Transmission of voice over such network makes it more applicable in real world. This research work was devoted solely towards betterment of such feature. A protocol is also developed in this regard. But simulation test of the protocol is yet to be done. This protocol adapts to the dyanamiticity of ad-hoc realm to support service discovery of SIP Registrar in a distributed manner. It is dynamic in a sense that given value of cell diameter this protocol distributes the registrar servers evenly in the whole topology, but static in terms of protocol parameters. Different network scenarios are suitable for different protocol parameters. A possible improvement of this protocol can be dynamic cell diameter to better adaption to changing environment. Improving this protocol is not just the end of it. There are many more issues still untouched in voice communication on ad-hoc networks and improving those can have great effect on improving voice transmission quality on such network.

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