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A New Distributed Application and Network Layer Protocol for VoIP in Mobile Ad Hoc Networks

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ABSTRACT

Mobile Ad-hoc Networks (MANETs) are a kind of Wireless ad hoc network that usually has a routable networking environment on top of a Link Layer ad hoc network. The main aim of the project is to propose a new protocol for Voice over IP (VoIP) transmissions in wireless ad-hoc networks. Each device in a MANET is free to move independently in any direction, and will therefore change its links to other devices frequently. Each must forward traffic unrelated to its own use, and therefore be a router. The primary challenge in building a MANET is equipping each device to continuously maintain the information required to properly route traffic. Such networks may operate by themselves. They may contain one or multiple and different transceivers between nodes. This results in a highly dynamic, autonomous topology. VoIP over MANETs is a challenging issue due to the intrinsic distributed nature of the existing peer-to-peer paradigm. This paper proposes a new protocol, capable of ensuring a Quality of Service level for VoIP calls over a MANET and to manage calls in the system. Voice over IP(VoIP) is a methodology and group of technologies for the delivery of voice communication and multimedia sessions over internet protocol networks.

Keywords: optimizing path selection through distance vector routing, voice communication through ip.

I. INTRODUCTION

The main aim of the project is to propose a new protocol for Voice over IP (VoIP) transmissions in wireless ad-hoc networks**.** Distributed architecture is necessary when dealing with dynamic environments, such as ports or battlefields, where creating infrastructures becomes expensive or impossible. Mobile Ad-hoc Networks (MANETs) are based on a peer-to-peer approach and each node participates in the organization of the whole network. VoIP over MANETs is a challenging issue due to the intrinsic distributed nature of the existing peer-to-peer paradigm. This paper proposes a new protocol, capable of ensuring a **Quality of Service (QoS)** level for VoIP calls over a MANET and to manage a higher number of calls in the system. Novel metric and utility functions are proposed to perform the best path selection from source to destination nodes, respecting the QoS parameters for VoIP quality.

An analysis of currently used VoIP systems shows how they are characterized by a set of fixed nodes, which act as intermediaries between their endpoints or provide registration and localization of nodes. This kind of approach has some disadvantages that make it unsuitable when system dimensions grow:

- **Low fault-tolerance** (if a proxy is damaged, the whole system will not work).
- **Low scalability** in the number of supported parallel calls.

Traditional VoIP structures become inadequate, because of the need for different fixed nodes. If we consider a scenario with close endpoints and a far proxy then, without an infrastructure, the QoS levels become unacceptable: although a multi-hop protocol may forward the information to the proxy, the degradation introduced by each hop determines that constraints on the **Mean Opinion Score (MOS)** are not respected, with consequent low-quality admitted calls.

The proposed solution is based on a novel metric related to an objective measure of the QoS for calls and on optimal codec selection in the route discovery phase of the wireless routing protocols for MANET, Each node behaves actively and passively for routing operations. A Mobile Ad-hoc network is a wireless ad-hoc network which is used to exchange information. A distributed scenario has been considered each node can communicate with its neighbors. A neighbor has direct radio coverage in the considered environment. Each node behaves actively and passively for routing operations. Ad hoc on demand multi path distance vector routing protocol initiate route discovery only when a route is required. After finding the all path from source to destination, the destination find the short path or short hop then replay to the source. Managing construction of the lowest-cost path from source to destination in the best way on the basis of an objective measure of the Quality of service to make the call to destination using VoIP. VoIP technology is converting voice signals to data packages and transferring them on IP network.

The new functionalities of the proposed system are

1. Managing the construction of the lowest-cost path from source to destination in the best way on the basis of an objective measure of the QoS of the calls.
2. Dynamic codec selection strategy, in order to guarantee the best quality for new incoming calls, without degrading system performance.
3. Call admission control procedure integrated in the route selection, to refuse or direct on alternative paths
4. The additional calls that can degrade the VoIP QoS constraints;
5. Route selection based on a suitable flexibility index, which allows the system to maximize the number of admissible new calls with the available resources, and hence to scale with the network size increase.

II. METHODS AND MATERIAL

A. Network Formation

In network formation, we create a nodes and grouping all nodes. In node creation we have to give distance and range. Based on distance and range the node will be created. Each node sends "**hello**" message to other nodes which allows detecting it. Once a node detects "**hello**" message from another node (neighbor), it maintains a contact record to store information about the neighbor. Using multicast socket, all nodes are used to detect the neighbor nodes. Thus in this project we are establishing a connection between various nodes and thus by which we are forming the network atmosphere. The node which wants to share the information is acts as a source and the node which the source wants to communicate with it is acts as a destination. the communication between the source node and destination is processed through The neighbor nodes of each node. Each node in the network is assigned to have a distance and range by which the particular node can communicate.

B. Optimizing Path Selection

Determining routes requires accumulating the address of each node between the source and destination during route discovery. Source makes the request, the terminal broadcasts the packet. All neighbors will receive and forward request. Sooner the request will be received by the destination, which will set a timer for the receipt of other Challenge packets generated by the same source. When the timer expires, node destination will elaborate all received requests and will reply only to the best Challenge request. The one belonging to the path with the lowest cost.

Here the alternative path selection can be easily carried out as the algorithm that we are using here is the distance vector routing which is dynamic source routing algorithm. Distance vector routing algorithm keeps track of the periodical changes so that when the nodes go out of the range it is easy to find an alternative path.

C. Voice Communication using VOIP

VoIP stands for Voice over Internet Protocol. As the term says VoIP tries to let go voice (mainly human) through IP packets and, in definitive through Internet. VoIP works like that, digitalizing voice in data packets, sending them and reconverting them in voice at destination. A duplex communication system is a point to point system composed of two connected parties or devices that can communicate with one another in both directions.

In a half-duplex system, there are still two clearly defined paths, and each party can communicate to the other but not simultaneously; the communication is one direction at a time. In a full duplex system, both parties can communicate to the other simultaneously. We are choosing alternate path when there is any distortion or signal problem between two users. Distortion will be occurs based on mobility. In this case only dynamic path will choose. After finding dynamic path, communication will be performed securely. It is a continues process.

III. RESULTS AND DISCUSSION

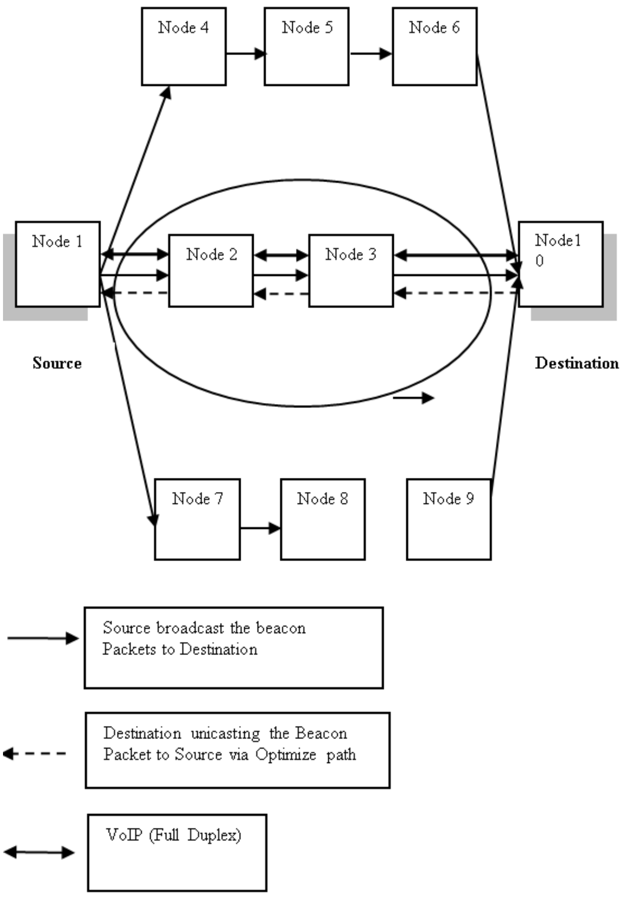

Figure 1: System Architecture

The above fig 1 is the architecture of our concept here the data's to be sent from the source to destination is processed through neighbor nodes and the it reaches the destination.

The performance of the wireless sensor network, to execute this project on LAN or wifi communication channel. So we need to one or more than machine to execute the demo. Machine needs the enough hard disk space to install the software and run our project.

IV. CONCLUSION

Thus in this project we are establishing a connection between various nodes and thus by which we are forming the network atmosphere. The node which wants to share the information is acts as a source and the node which the source wants to communicate with it is acts as a destination. The communication between the source node and destination is processed through the neighbor nodes of each node. Each node in the network is assigned to have a distance and range by which the particular node can communicate. The performance of the wireless sensor network, to execute this project on LAN or Wi-Fi communication channel.

In this concept if any node goes out of the range and if that particular node is responsible for the communication between the source and destination, then the system choses the alternative path between the source and destination and choses optimized path. Through this optimized path communication is continued without any distortion and here the communication is peer to peer communication by which both the source and the destination can actively participate in system.

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