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VoIP QoS Prediction over Wireless Mesh Network Scenario

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Abstract: Quality of Service (QoS) made its requirement necessary for the upgradation to next generation wireless networks which can provide support over multimedia network applications, say for example Voice over Internet Protocol (VoIP). With Wireless Mesh Networking, a visionary future solution, bagged with promising technology stack can be seen for next networks generations which will be providing support to VoIP application. VoIP has served to be a beneficial fit in the case of Multimedia network application that has been tested over emerging wireless mesh network areas. However, the challenges for the same has made a long list. This Paper discusses E-Model which is the most reliable method for evaluating the voice quality defined by the International Telecommunication Union-Telecommunication (ITU-T). This paper analyzes voice quality in terms of R-Factor and MOS in IEEE 802.11s Wireless Mesh Network (WMN) and will give an overview of various codecs used in voice transmission. This paper also analyzes various Reactive and proactive routing protocols used in wireless mesh network.

Keywords: Wireless Mesh Network, Voice over IP (VoIP), Mean Opinion Score (MoS). R-Factor.

1. WIRELESS MESH NETWORK

Wireless Mesh Network (WMN) has recently emerged as an important Research area worldwide. They are dynamically self-organized and self-configured, with automatic establishment of an ad hoc network and maintenance of mesh connectivity done by the nodes in the network. There are two types of nodes in WMN i.e. Mesh routers and Mesh clients. Mesh routers or Mesh Points have minimal mobility i.e. they work without any energy constraints and form the mesh backbone for mesh clients[1]. However, the mesh clients can be either stationary or mobile node. Properties like low-cost and relative ease of deployment has made WMN an attractive communication paradigm. WMNs typically consist of many base stations, some of which are directly connected to the Internet. The users connect to one of the base stations, and the base stations form a multi-hop wireless network to route traffic between the Internet and the users. The routers enable conventional nodes equipped with wireless network interface cards (NICs) to connect directly to WMNs[2]. In the absence of wireless NICSs, Ethernet serves the purpose of connecting to wireless mesh routers, thus accessing WMNs. WMN caters to the need of the users to be always on line anywhere, anytime [3]. Instead of being another type of ad -hoc networking, WMNs diversify and enhance the capabilities of ad-hoc networks. In many ways WMNs have become preferable over MANETs, they have advantages such as low installation costs, easy network maintenance, robustness, service coverage that can be relied on, etc [4]. Today, WMNs are a widely accepted technology in the traditional application areas of ad hoc networks, and they are also undergoing rapid commercialization application scenarios such as broadband home networking, community networking, building automation, high-speed metropolitan area networks, and enterprise networking etc. The number of hops refer to the number of mesh routers data must pass through. Traffic that passes through five hops is transmitted five times over the wireless medium between devices. Single hop traffic is transmitted only one time over the wireless medium. The number of hops the data will pass through are dictated by the location of wireless station, the destination node, and the configuration of the WMN. Based on its network topology, wireless mesh networks are classified into three main groups i.e. Flat WMN, Hierarchical WMN, and Hybrid WMN[5].

Flat Wireless Mesh Network

In a Flat WMN, the network is formed by nodes that are both clients and routers. Here, each node is at the same level as that of its peers. Its downside include lack of network scalability and high resource constraints, however one of its advantages is its simplicity.

Hierarchical Wireless Mesh Network

In a hierarchical WMN, the network has multiple tiers or levels where the WMN client nodes form the lowest tier in the hierarchy. These client nodes can communicate with a WMN backbone network formed by WMN routers.

Hybrid Wireless Mesh Network

Hybrid WMNs are a special case of hierarchical WMNs where the WMN utilizes other wireless networks for communication. For example, Hybrid WMN uses other infrastructure-based WMNs such as cellular networks, WiMAX networks, or satellite networks.

2. VOIP ARCHITECTURE OVER WLAN

Voice over Internet Protocol (VoIP) [6] is a technology that transports voice data packets across packet switched networks using the Internet Protocol. It involves digitization of voice streams and transmitting the digital voice as packets over conventional IP-based packet networks like the Internet, Local Area Network (LAN) or wireless LAN (WLAN).Although the quality of VoIP does not yet match the quality of a circuit-switched telephone network. In WLAN, as VoIP technology is still in the early stages of commercial deployment, it is necessary to examine if VoIP over WLAN can provide a Quality of Service (QoS) comparable to that of the existing PSTN and cellular networks. There is a need of research efforts for investigating the QoS of VoIP amid the increasing popularity of the 802.11 based WMN .

ITU has given various standards which help in evaluating the quality of voice. ITU has recommended both subjective and objective evaluation methods for evaluating the quality of voice [7], [8]. One standard defines the quality of voice as tested by various listeners.
The parameter which defines the opinion of these listeners is Mean Opinion Score (MoS).The other standard involves measurement of the QoS Metrics. Objective evaluation methods are machine based; the voice quality is calculated using quantitative distortion parameters between the source and the destination endpoint.

3. E-MODEL

E-Model [7] defines various evaluation parameters for finding the quality of voice. The parameters include R-Factor and MoS. The formula for R-factor considers delay, packet loss, and type of compression algorithm. The formula for the R-Factor [7] is given in equation (1):

$$R = R_0 - I_e - I_s - I_d + A_d \qquad (1)$$

Here, R_0 represents quality of voice without any impairment, I_s represents impairment due to packet loss, I_d refers to loss due to jitter, I_e shows loss due to encoding and A_d is the Advantage Factor.

The other parameter which can be used for calculating the quality of VoIP is Mean opinion Score[6].It is recommended by ITU-T for representing the subjective parameter for the quality of VoIP. In finding Mean Opinion Score, various people rate the performance of the voice from 1 to 5. On the basis of their rating, voice quality can be categorized as Good, Average or Poor. Mean Opinion Score makes it very easy to make comparison between the different voice streams. The main limitation is the hiring of so many listeners for listening the sentences. There are also various other objective evaluation parameters which can be used for finding the quality of voice. E-Model [7] is recommended by ITU-T which gives call performance based on various factors in network. E -model takes into account a wide range of impairments, such as CODEC choice, end-to -end delay, packet loss and jitter. With these inputs, the E-model produces a rating factor, R. This rating factor can be transformed to give an estimated MOS value, which we use for measuring call quality in our experiments. One drawback of any call quality calculation such as the E-model is that it is only an estimate of sound quality. It does not test actual sound quality, but estimates are based on the performance metrics measured on the network. In voice communications, particularly Internet telephony, the Mean Opinion Score (MoS) provides an aggregate numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinion scores) that are mathematically averaged to obtain a quantitative indicator of overall quality [8].

There are various technical factors like echo, delay and distortion which can be calculated by various methods but the subjective factors cannot be calculated by the technology, so there emerges a requirement of MoS. Both R- factor and MoS are interdependent on each other.
Upto now, very little research work has been done for investigating the performance of VoIP over WMNs.The quality of a VoIP Call depends upon the Routing protocols i.e. reactive or proactive required to route voice packets, routing metrics, the type of codec used and the signaling protocol used.

4. ENCODING ALGORITHMS FOR VOIP

There are a number of codecs exist in real life, but mainly codecs G.711,G.723.1 and G.729 are mainly utilized for encoding the voice traffic.G.711 is the traditional codec which requires high bandwidth for transmission of voice. Requirement of high bandwidth no doubt makes the voice quality to be optimal but in some application, a much lower bit rate is required for carrying the high amount of traffic within limited capacity. But in some previous years many other new encoding algorithms like G.723.1 and G.729 have been developed which affect the quality metrics of the Voice traffic defined by E-Model. Different research works have been carried for selection of appropriate codec for different networks. Codecs perform according to the environment i.e.it depends upon the network architecture of wired, wireless LAN or WiMax and users density[9].

G.711 is the the traditional codec which is used in Public Switching Telephony Network which does not perform any compression. It uses the technique of Pulse code modulation (PCM) It generates 64 kbps stream, with low CPU Utilization .It generates best audio quality as compared to other codecs due to uncompressing of voice signal. The main limitation of G.711 is that it uses high bandwidth than other codecs [9] .It sends 160 byte payload at bit rate of 64 kbps [9],[10].

G.723.1 codec belongs to the category of G.723 codec. It transfers the voice signal at bit rate of 5.3 Kbps. This codec utilizes the technique of ACELP i.e. Algebraic code excited linear prediction Algorithm .The other codec G.723.a works at 6.3 Kbps bit rate. It provides good voice quality in terms of parameters like packet loss and bit errors.

G. 729 codec is a licensed compression technique which is designed to provide good voice call quality without any high consumption of bandwidth[10].Due to compression of signal, It uses less bandwidth than G.711.It follows the procedure on CS-ACELP technique i.e. Conjugate-Structure Algebraic-Code-Excited Linear Prediction algorithm with 8 kbps bit rate.

5. MESH ROUTING PROTOCOLS

The various routing protocols which can be used for communication of VoIP traffic in wireless mesh network can be proactive (e.g. DSDV, OLSR), Source initiated Reactive Routing protocols (e.g. AODV, DSR) and Hybrid Routing Protocols (e.g. HWMP, HSLS)

Adhoc on Demand Distance Vector (AODV)

AODV [11] is a reactive routing protocol used as an on demand routing protocol. It design routes as per demand or as desired by the starting node. Request (RREQ) is transferred to every node from the sender so that route can be established. On getting RREQ, the route reply i.e. RREP is generated and is transferred back to the sender. If any link fails or there occurs any type of error, Route error reply i.e. RERR is generated. After receiving RERR, source node can again start with the route discovery process. The main advantage of AODV is that it creates routes only when source has some packets to send. The route gets inactive when source does not have any data to send. As the routes are created when needed, it allows the nodes to enter or leave the network at their own will. It uses sequence numbers to ensure that there are no loops while routing. Every node maintains a sequence number which increases monotonically every time node notices change in neighboring nodes.

Dynamic Source Routing (DSR)

The Dynamic Source Routing protocol (DSR)[12] is basically used in multi-hop wireless ad hoc networks of mobile nodes. When node (Source) wants to send a packet to another node (Destination), but does not know the route, it initiates a route discovery, also called On-demand or reactive routing. Every node contains a Routing Database in which it maintains a complete route to recent destinations in the form of list. If some required destination is on the list, it just uses that route from the list to reach the destination, if not, it broadcasts a Route Request (RREQ) to all neighbors. DSR has the same working as AODV i.e. it makes a route on demand request while making any communication. But it mainly uses source routing rather than routing table lying at every node.

Destination-Sequenced Distance-Vector Routing (DSDV)

DSDV [12] is an algorithm based on Bellman Ford algorithm. It is a table driven routing protocol which is developed by C. Perkins and P.Bhagwat.In DSDV, nodes require regular update information related to routing tables. DSDV requires a regular update of its routing tables, which uses up battery power and a small amount of bandwidth even when the network is idle. Whenever the topology of the network changes, a new sequence number is necessary before the network re-converges; thus, DSDV is not suitable for highly dynamic networks.

Optimize Link State Routing (OLSR)

This is a proactive routing protocol which actually maintains data about topology of the network[13] .Every node in the network sends messages titled “HELLO” with predefined intervals. OLSR uses Multipoint Relays (MPRs) for sending the control traffic to the neighboring nodes.

6. CONCLUSION

This paper has reviewed the concept of Wireless Mesh Network, types and its applications. ITU-T’s E-model which is used in speech quality prediction in VoIP scenarios based on various Parameters i.e. R factor and MOS is analyzed. Also various codecs (G.711, G.723, and G.729) and Routing Protocols which affect the quality of voice are discussed.

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