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Analysis of VoIP E-Model and Routing Protocols over 802.11 Wireless Mesh Network

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Abstract— **For next generation wireless networks, supporting quality of service (QoS) in multimedia application like Voice over IP is a necessary and critical requirement. Wireless Mesh Networking is envisioned as a solution for next networks generation and a promising technology for supporting VoIP application. VoIP has become a killer application and is gradually being tested over emerging areas like Wireless mesh networks. There are various challenges for VoIP in WMN. 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). This Paper gives 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), R-Factor, Mean Opinion Score (MOS).

I. INTRODUCTION

Wireless Mesh Networks have recently emerged as an important Research area worldwide. Wireless mesh networks (WMNs) are dynamically self-organized and self-configured, with the nodes in the network automatically establishing an ad hoc network and maintaining the mesh connectivity. WMNs are comprised of two types of nodes 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. However, the mesh clients can be either stationary or mobile node. Wireless mesh networks are an attractive communication paradigm because of their low cost and relative ease of deployment. 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. Wireless mesh routers enable conventional nodes equipped with wireless network interface cards (NICs) to connect directly to WMNs [1][2]. Ethernet can be used to access WMNs by connecting to wireless mesh routers when wireless NICs are not available. 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 [5].With WMNs, the number of hops refers 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 location of a wireless station, the destination node, and the configuration of the WMN dictate the number of hops the data will pass through.

Based on its network topology, wireless mesh networks are classified into three main groups i.e. Flat WMN, Hierarchical WMN, and Hybrid WMN

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. One of its advantages is its simplicity, and its disadvantages include lack of network scalability and high resource constraints.

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

II. VOIP ARCHITECTURE

Voice over Internet Protocol (VoIP) [8],[9] is a technology that transports voice data packets across packet switched networks using the Internet Protocol. VoIP 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) [5],[6].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. Also as the 802.11 based WMNs is gaining popularity, the research efforts are required to investigate the Quality of Service of VoIP over such multi hop networks. Figure 1 shows the basic VOIP architecture -

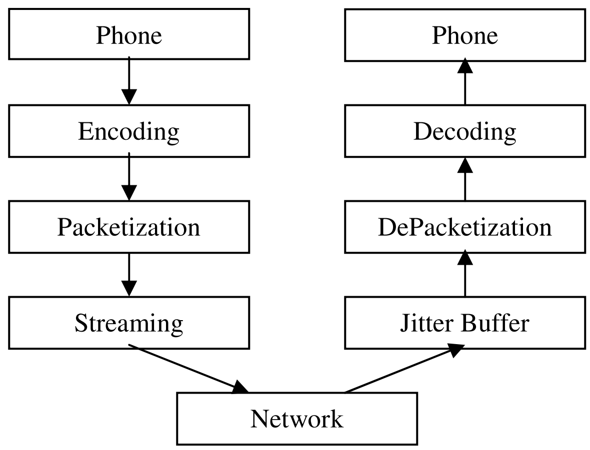


Fig.1: VoIP Architecture

In a typical VoIP application, a voice signal is sampled, digitized, and encoded using a given algorithm/coder. The encoded data is packetized and transmitted using RTP/UDP/IP [14]. At the receiver's side, data is de-packetized and forwarded to a jitter buffer, which smooths out the delay incurred in the network. Finally, the data is decoded and the voice signal is reconstructed. In a VoIP system, the total mouth-to-ear delay is composed of three components: codec delay, jitter delay, and network delay. However, WMNs are multi-hop in nature and add additional delay when VoIP traffic is passed over them. Also the delay added by WMN depends on the number of wireless hops over which the traffic has travelled.

III. VOIP CODECS

A codec (coder/decoder) [10] converts an analog signal to a compressed digital bit stream, and another identical codec at the other end of the communication converts the digital bit stream back into an analog signal. In a VoIP system, the codec used is often referred to as the encoding method. Codecs generally provide a compression capability to save network bandwidth. Some codecs also support silence suppression, where silence is not encoded or transmitted. The QoS on VoIP network partly depends on the types of voice codec used [11]. The primary functions of a voice codec are to perform analog/digital voice signal conversion and digital compression. Among three commonly used codec in Internet telephony are G.711, G.723.1, and G.729. These codecs differ in their coding rate (bps), frame rate (frames/s), algorithmic latency that will influence the speech quality or Mean Opinion Source (MOS) [12] in a VoIP network.
Payload size for each codec depends on the codec speed or data rate. The G.711 has speed of 64 Kbps and if each speech packet size is of 20 ms, then the payload size for G.711 will be of 160 bytes.

Payload Size (bytes) = [codec speed (bits/sec)*speech packet size (ms)] / [8(bits/byte) * 1000(ms/sec)]

Payload size of 160 bytes for G.711 codec means that the codec produces 160 bytes chunks of VoIP traffic every 20 ms interval. The G.711 codec gives the best voice quality, since it performs no compression, introduces the least delay, and is less sensitive than other codecs to packet loss. Other codecs, like G.729 and the G.723 consume less bandwidth by compressing the signal. In this research work, G.711 codec has been used because of its good voice quality and least delay. Table 1 shows various attributes of codecs.
G.711 codec-In wireless networks, G.711 is applied for encoding telephone audio signal at a rate of 64 kbps with a sample rate of 8 kHz and 8 bits per sample. In an IP network, voice is converted into packets with durations of 5, 10 or 20ms of sampled voice, and these samples are encapsulated in a VoIP packet.

G.723m / G.723a codec-It belongs to the Algebraic Code Excited Linear Prediction (ACELP) family of codec and has two bit rates associated with it: 5.3 Kbps and 6.3 Kbps. The encoder functionality includes Voice Activity Detection and Comfort Noise Generation (VAD/CNG) and decoder is capable of accepting silence frames. The coder operates on speech frames of 30ms corresponding to 240 samples at a sampling rate of 8000 samples/s and the total algorithmic delay is 37.5ms. The codec offers good speech quality in network impairments such as frame loss and bit errors and is suitable for applications such as VoIP.

G.729 codec-The codec belongs to the Code Excited Linear Prediction coding (CELP) model speech coders and uses Conjugate Structure - Algebraic Code Excited Linear Prediction (CS-ACELP). This coder was originally designed for wireless applications at fixed 8 kbit/s output rate, not including the channel coding.

Table 1: Different Attributes of Codec

Codec	Data Rate (kbps)	Speech Packet Size (ms)
G.711	64.0	20 ms
G.729	8.0	20 ms
G.723m	6.3	30 ms
G.723a	5.3	30 ms

Table 2: Different Attributes of Codecs

IV E-Model Review

The method defined by the E-Model [11],[12] derives the R-Factor as to find the Quality of voice. It does not only take in account transport delay and network packet loss, but it also considers the voice application characteristics, like the codec quality, codec robustness against packet loss. Thus delay, delay jitter and packet loss have been integrated in single parameter R.

$$R = R_0 - Id - Ie - Is + A$$

R_0 is voice quality without distortion. Usually it is equal to 100.
Id corresponds to impairment level caused by delay and delay jitter.
Ie represents the impairment caused by encoding artifacts.

Is represents impairments caused due to echo and packet loss.
A is Advantage Factor that a user can tolerate to decrease voice quality.For instance, the A value is greater in satellite networks than in classical circuit-switched networks, because user expectations in satellite networks are lower than those in wired networks. The typical range for the A factor is [0, 20]. Since human being can tolerate a wide range of these parameters and different persons have different expectation of voice quality. Figure 2 shows the dependence of R Score on various parameters.

To measure call performance, a value known as the MOS is used. This is an industry standard number, detailed in ITU-T recommendation that represents perceived call quality ranges from 1 to 5.The MOS is a representation of the quality of human speech. To determine MOS for a specific configuration, a number of listeners rate the quality of test sentences read by both male and female speakers. Each sentence is given a rating, from 1 to 5, 1 being the worst, and 5 being the best. The MOS of a specific configuration is the arithmetic mean of the individual MOS values as recorded by the listeners.

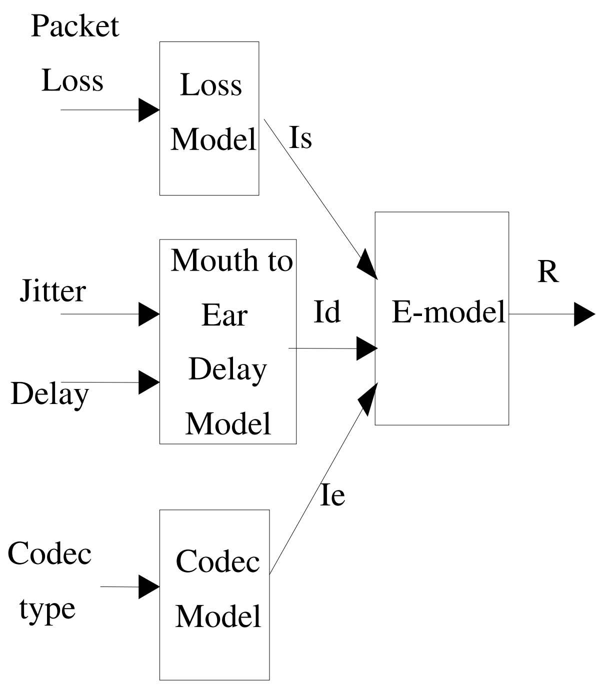


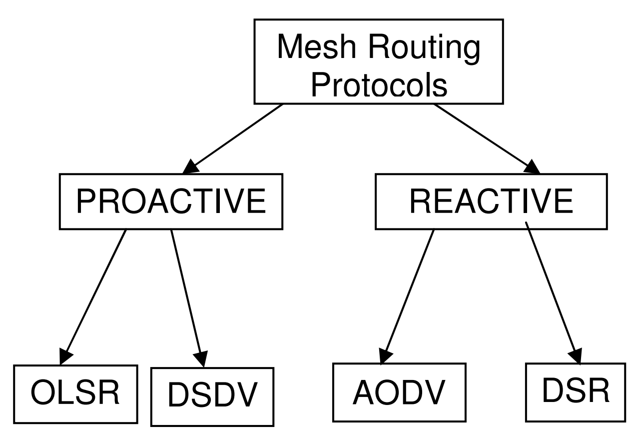
Fig 2: Dependence of R Score on various parameters

Typical MOS values of a cell phone call are in the upper-3 range while a land phone line ranks in the mid-4 range. MOS is a very useful means of measuring voice quality, as it allows for easy comparison of voice call quality from one test to the next. The drawback with MOS, however, is the cost of hiring many people to listen to sentences and rate them.
The E-model is a recommendation published by the ITU-T, given recommendation identifier G.107. It is a transmission rating model that gives an estimated call quality based on network factors. The most recent version of the E-model was approved by the ITU-T in the year 2005.

We chose to use the E-model because it takes into account a wide range of impairments, such as CODEC choice, end-to-end delay, packet loss and jitter. Given 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.
Technical quality like loss, delay, distortion, noise and echo can be measured by the technology. But subjective measures of quality are not possible to measure by the technology and for that we need MOS. To determine MOS, a number of listeners rate the quality of test sentences read aloud over the communications circuit by male and female speakers. A listener gives each sentence a rating.
MOS is usually measured by a panel of human testers.
MOS value can be calculated with following formula.
$MOS = 1 + 0.035R + 7 \times 10^{-6} \times R(R - 60)(100 - R)$
The MOS is the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best) [12]
(1) Bad (2) Poor (3) Fair (4) Good (5) Excellent
VoIP has become killer application in WMN [17]. To date few research works has been carried out to investigate the performance of VoIP over WMNs [15-19], and many challenging issues are remain to be resolved. One of the issues is the effect of increase of hops on quality of the voice in a wireless mesh network which we have discussed in this research work.
For a VoIP call [13], the most critical requirement is QoS. For a network delivering VoIP calls, critical requirement is that VoIP calls can co-exist with other traffic and also, given a certain traffic load of other traffic, the number of VoIP calls needs to be as high as possible.

V. MESH ROUTING PROTOCOLS

Routing Protocols used in wireless mesh network can be classified into two main categories: Proactive or table driven routing protocols and Reactive or on-demand routing protocols [7] shown in Figure 3.



Ad hoc On Demand Distance Vector (AODV)

The Adhoc on Demand Distance Vector (AODV) routing algorithm [20] is a routing protocol designed for ad hoc mobile networks. AODV is capable of both unicast and multicast routing. It is an on demand algorithm, meaning that it builds routes between nodes only as desired by source nodes. It maintains these routes as long as they are needed by the sources. Additionally, AODV forms trees which connect multicast group members. The trees are composed of the group members and the nodes needed to connect the members. AODV uses sequence numbers to ensure the freshness of routes.

It is loop-free, self-starting, and scales to large numbers of mobile nodes.AODV build routes using a route request / route reply query cycle. When a source node desires a route to a destination for which it does not already have a route, it broadcasts a route request (RREQ) packet across the network. Nodes receiving this packet update their information for the source node and set up backwards pointers to the source node in the route tables. In addition to the source node's IP address, current sequence number, and broadcast ID, the RREQ also contains the most recent sequence number for the destination of which the source node is aware. A node receiving the RREQ may send a route reply (RREP) if it is either the destination or provides a route to the destination with corresponding sequence number greater than or equal to that contained in the RREQ. If this is the case, it unicasts a RREP back to the source. Otherwise, it rebroadcasts the RREQ. Nodes keep track of the RREQ's source IP address and broadcast ID. If they receive a RREQ which they have already processed, they discard the RREQ and do not forward it. As the RREP propagates back to the source, nodes set up forward pointers to the destination. Once the source node receives the RREP, it may begin to forward data packets to the destination. If the source later receives a RREP containing a greater sequence number or contains the same sequence number with a smaller hop count, it may update its routing information for that destination and begin using the better route. Once the source stops sending data packets, the links will time out and eventually be deleted from the intermediate node routing tables. If a link break occurs while the route is active, the node upstream of the break propagates a route error (RERR) message to the source node to inform it of the now unreachable destination(s). After receiving the RERR, if the source node still desires the route, it can reinitiate route discovery.

Dynamic Source Routing (DSR)

Dynamic Source Routing (DSR) [23] is a routing protocol for wireless mesh networks. It is similar to AODV in that it forms a route on-demand when a transmitting computer requests one. However, it uses source routing instead of relying on the routing table at each intermediatedevice. Many successive refinements have been made to DSR, including DSRFLOW. Determining source routes requires accumulating the address of each device between the source and destination during route discovery. The accumulated path information is cached by nodes processing the route discovery packets. The learned paths are used to route packets. To accomplish source routing, the routed packets contain the address of each device the packet will traverse. This may result in high overhead for long paths or large addresses, like IPv6. To avoid using source routing, DSR optionally defines a flow id option that allows packets to be forwarded on a hop-by-hop basis. This protocol is truly based on source routing whereby allthe routing information is maintained (continually updated) at mobile nodes. It has only 2 major phases which are Route Discovery and Route Maintenance.Route Reply would only be generated if the message has reached the intended destination node (route record which is initially contained in Route Request would be inserted into the Route Reply). To return the Route Reply, the destination node must have a route to the source node. If the route is in the Destination Node's route cache, the route would be used. Otherwise, the node will reverse the route based on the route record in the Route Reply message header (symmetric links). In the event of fatal transmission, the Route Maintenance Phase is initiated whereby the Route Error packets are generated at a node. The erroneous hop will be removed from the node's route cache, all routes containing the hop are truncated at that point. Again, the Route Discovery Phase is initiated to determine the most viable route.

Destination-Sequenced Distance-Vector Routing (DSDV)

Destination-Sequenced Distance-Vector Routing (DSDV) [21] is a table-driven routing scheme for ad hoc mobile networks based on the Bellman-Ford algorithm. It was developed by C. Perkins and P.Bhagwat in 1994. The main contribution of the algorithm was to solve the Routing Loop problem. Each entry in the routing table contains a sequence number, the sequence numbers are generally even if a link is present; else, an odd number is used. The number is generated by the destination, and the emitter needs to send out the next update with this number. Routing information is distributed between nodes sending full dumps infrequently and smaller incremental updates more frequently. DSDV was one of the early algorithms available. It is quite suitable for creating ad hoc networks with small number of nodes. Since no formal specification of this algorithm is present there is no commercial implementation of this algorithm. Many improved forms of this algorithm have been suggested. 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)

OLSR [22] is a proactive link state routing protocol. As a proactive protocol, OLSR constructs and constantly maintains information about network topology by means of exchange link state information. Each OLSR node sends HELLO messages in predefined time intervals for constructing its 1-hop and 2-hop neighbour sets and a TC (topology control) message for completing link state information, so routing table can be calculated. Link failures in OLSR are detected this way. OLSR introduces multipoint relays (MPRs) in order to reduce message overhead in network. The MPR set of a given OLSR node is a subset of its neighbours which can forward its control messages. The neighbours which a given node A selects as MPR are called MPR nodes of A. When all neighbours are MPR nodes of a given router, OLSR diffuses control messages similarly to classical flooding mechanism. On the other hand, MPR mechanism described in can decrease network performance due overhead introduced for constructing and repairing MPR set. If one or more MPR nodes fail, link state information cannot be completely diffused, thus some routers can forward user data by invalid paths on network.

VI. CONCLUSION

This paper has reviewed the concept of Wireless Mesh Network, its advantages 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) which affect the quality of voice are discussed.

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