Journal of Applied Sciences

ISSN 1812-5654

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Journal of Applied Sciences 7 (15): 2111-2119, 2007
ISSN 1812-5654

Performance Evaluation of Mobile Ad Hoc Network Based Communications for Future Mobile Tele-Emergency System

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Abstract: Sparked by awareness of the limitations to provide medical services in remote areas, researchers have perceived that developing telemedicine systems is inevitable. In most cases very remote areas and disaster struck areas lack telecommunication infrastructure. Telemedicine system operating in such areas must have advanced wireless technology supporting it in devastating situation, hence it is called as tele-emergency system. Our approach is on MANET combined with Mobile IP and MIPV6, is the basis of infrastructure for the mobile tele-emergency system. The tele-emergency system requires data, voice and video transmission in its network. In this investigation, evaluation is based on simulation of the various ITU-T standard CODECs of VoIP and video transmission over MANET using discrete event simulator NS-2. The results of simulation showed that ITU-T G723.1 worked well in the MANET environment than the other CODECs for VoIP in fixed and mobility tele-emergency environments. From the simulation of video CODEC performance evaluation, it was observed that H.263 performed to a great extent in random small scale environment and also in multiple video flow of 57.6kbps speed video transmission.

Key words: Mobile ad-hoc network (MANET), mobile IP, WLAN, telemedicine, ad-hoc on demand distance vector (AODV)

INTRODUCTION

Telemedicine is defined as the delivery of medical health care and medical expertise using a combination of telecommunications technologies. The tele-emergency system is defined as the delivery of health care and sharing of medical knowledge over a distance using the telecommunication applications in emergency situations. Telemedicine systems can support applications ranging from videoconference to providing diagnostics, high quality image and still-image and medical database record. The Tele-emergency project proposed tele-emergency units as mobile telemedicine units. The system is based on three technologies viz., MIPv6, MANET and WLAN. The integration of the technologies will produce a highly capable system with the ability to be rapidly deployed to support medical services. The integration of MANET with Mobile IP has been introduced (Lamont *et al*., 2003; Lamont *et al*., 2002; Tseng *et al*., 2003).

Tele-Emergency requirements and applications: The requirements of the Tele-Emergency are:

- Capable to work in remote areas, which has limited communications infrastructure.
- Capable of being deployed for emergency condition and managing electronic patient records.
- Supported by real-time multimedia communications and Geographical Information System (GIS).

Low operating cost: Applications in the Tele-Emergency are classified into basic and extended services. Basic services applications are digital electrocardiogram (ECG), oxy-meter (spo2 meter), patient database record and location information based on GPS technologies. Extended services applications are complete multimedia services. All services can be used in rural areas based on wireless communication despite hospitals, which have wired communication.

SYSTEMS AND NETWORK ARCHITECTURE

Mobile ad hoc network: Ad hoc literally means, formed or used for specific or immediate problems or needs. Thus, MANET means a mobile network, which can be formed or used for specific or immediate problems or needs.

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Mobile nodes in a MANET communicate to each other without base station, without the aid of any centralized administration hence it is also known as an infrastructure less wireless network. MANET employs its mobile nodes as a part of the networking system. Each node in MANET can act as an intermediate node, i.e. as a relay to forward packets of data (Toh, 2002) and do routing functionality. In MANET, mobile nodes are free to move arbitrarily. It leads to an important property of MANET, which is a dynamic topology.

MANET routing protocols can be classified into demand-driven routing protocols and table-driven routing protocols. Demand-driven protocols create routes only when the source node initiates a route discovery process. Examples of demand-driven protocols are Ad Hoc On-Demand Distance Vector (AODV) (Perkins, 1999) and Dynamic Source Routing (DSR) (Johnson and Maltz,1996). Table-driven protocols attempt to maintain consistent, up to date routing information in routing tables on every node. Examples of table-driven protocols are Destination Sequenced Distance Vector (DSDV) (Perkins and Bhagwat, 1994) and Optimized Link State Routing (OLSR), (Jacquet *et al*., 2001).

Mobile IP: The traditional way of IP address assignment to a node is network dependent. It brings problem in a mobile network environment. When a mobile node moves from one wireless network to other network, the IP address must be changed accordingly, while ongoing connection must be maintained and the packets belonging to the connection must be delivered continuously. Mobile IP is the solution to this problem. Mobile IP users keep the same long-term IP address, i.e., home address, which has the same network prefix as a network called home network. When a Mobile Node determines that it is connected to a foreign network, it acquires a care-of address in addition to its home address. Care-of address is a forwarding address for a roaming mobile node.

In mobile IP, packets destined for the mobile node are sent to the mobile's home network. When a mobile node moves to a foreign network, it gets the care-of address from a router in the foreign network, called foreign agent. The mobile node then registers the new location to a router in its home network, called home agent. The home agent captures packets meant for the roaming mobile node, encapsulates and forwards it to the foreign agent. The foreign agent then delivers the packets to the mobile node. Packets in the reverse direction from the mobile node can go directly to the corresponding host without going through the home agent.

Mobile IPv6 simplifies the scenario by removing the foreign agent. The mobile node uses IPv6 address auto-configuration procedure to acquire a collocated care-of address (Perkins and Johnson, 1996; Johnson *et al*., 2002).

Network architecture: The Tele-Emergency network consists of at least one sub network using 2.4 GHz WLAN MANET. Figure 1 shows two MANET based sub

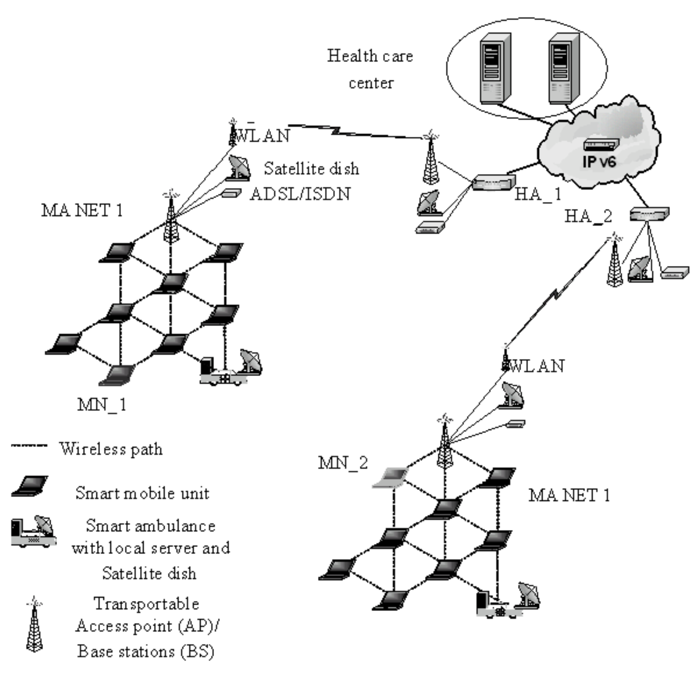

Fig. 1: System design

networks. In a sub network, Mobile Nodes (MNs) communicate directly with one another in a peer-to-peer connection and each MN acts as a router for any other nodes.

The health practitioners use MNs to transmit patient's data to the health care center or mobile ambulance. The mobile ambulance is a Tele-Emergency MN, which is equipped with a local server, several Tele-Emergency MNs and some optional communication devices such as satellite dish, ADSL and ISDN modem. The server in the mobile ambulance functions as a local database when there is no connection to the health care center. However, the optional satellite communication link and wired terrestrial communication in Fig.1 are used only if the field condition is making them more feasible than using 2.4 GHz WLAN.

The health care center, e.g., hospitals, has its own wireless LAN based medical systems working in the 2.4 GHz ISM band (Konstantinos *et al*., 2000).

Tele-emergency's mobility with MIPv6 and MANET: Every mobile node can route packets within a sub network based on a MANET routing protocol. Since mobile nodes can move arbitrarily, the network topology can change according to the ongoing moves. In Fig. 1, MN_1 is a member of sub network MANET 1. This sub network is the home network of MN_1. The other network, sub network MANET 2, is MN_2's home network and a foreign network for MN_1. The Mns use MIPv6 for addressing. Each Tele-Emergency mobile node (MN) has a home address (HoA) given by the home agent (HA) in the home network. The home network has a network prefix matching that of the MN's home address. When the mobile node is away from home network, it uses the address auto-configuration procedure defined in IPv6 to get a care-of-address (CoA).

When a MN sends packets to another node within the same sub network, the packets will only have to go through a MANET routing protocol. When MN_1 sends packets to MN_2, which is in a different sub network, the packets go through different types of routing. First, they are sent from MN_1 to the transportable base station in sub network 1 by using a MANET routing protocol, which then forward them to HA_1. HA_1 sends the packets to HA_2 by using IPv6 routing. HA_2 then forwards the packets to the transportable base station in sub network 2. Finally the packets again have to be routed by a MANET routing protocol to go to MN_2. A mobile node such as MN_1 can move anywhere, anytime in the home network or move into foreign network as shown in Fig. 2. When MN_1 is attached to a foreign network, it obtains a care of address (CoA) and registers it with HA_1. The HA_1 will know the current address and location of the MN_1.

When a Correspondent Node (CN) such as the server in health care center sends packets towards MN_1's home address, the packets are intercepted in MN_1's home network by HA_1 and tunneled to MN_1's care-of

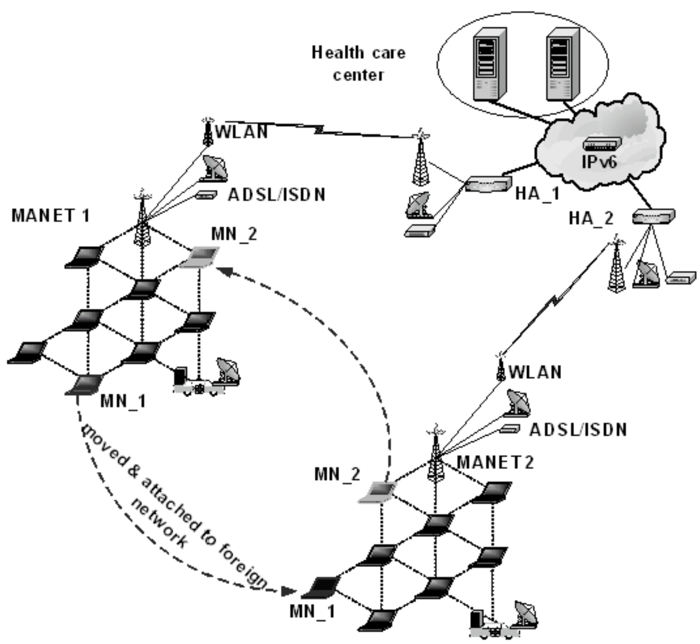

Fig. 2: Tele-Emergency mobility with MIPv6

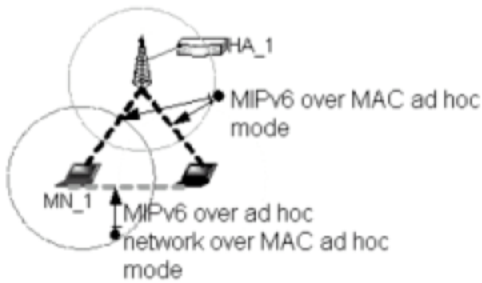

Fig. 3: Simple connection of telemedicine network

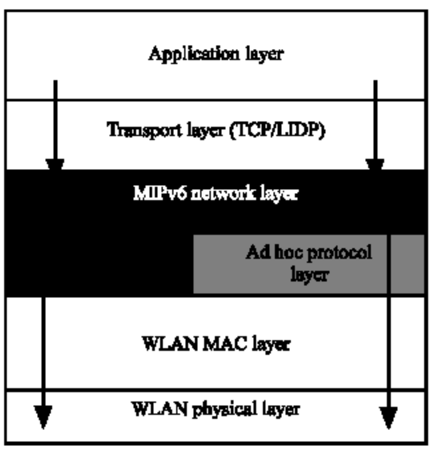

Fig. 4: Telemedicine MN protocol architecture

address. When MN_1 sends packets to the CN, it uses its CoA as source address and puts its HoA in the Destination Options (home address options). MN_1 informs the CN of its new CoA so that subsequent data traffic can be sent directly between MN_1 and CN, without tunneling process to the HA_1. Upon the packet receive by CN; the HoA replaces the source address so that the applications in CN perceive that it is still communicating with the MN at its HoA. While the CN sends packets to MN_1, it puts MN_1's CoA in the destination field and the HoA in the type 2 routing header. The same processes are applied to MN_2 when it is moved and attached into a different sub networks.

The macro mobility given by MIPv6 combined with MANET micro mobility within sub network provides seamless mobility for Tele-Emergency MN.

Telemedicine protocol architecture*:* In order to discuss about the protocol architecture, we need to examine the following case. When MN_1 is attached to its home network and located within coverage area of the AP, it has one hop to HA_1. MN_1 can use MIPv6 mechanism available from HA_1 to communicate with another node.

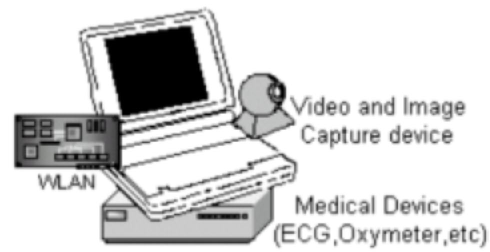

Fig. 5: Illustration of a smart mobile node

In the mean time, MN_1 can also communicate directly to another node in a peer-to-peer fashion using ad hoc network based on Medium Access Control (MAC) WLAN ad hoc mode. Therefore, that MN has two mechanism links over MAC WLAN ad hoc mode, namely MIPv6 over ad hoc mode and MIPv6 over ad hoc network. Figure 3 illustrates a simple connection between two MNs and a HA.

The telemedicine MN specifications are defined in a protocol architecture describing the functionality of all layers. As shown in Fig. 4, the telemedicine protocol architecture includes ad hoc protocol network within the network layer. If a MN communicates using ad hoc network, the ad hoc network layer is taken to provide ad hoc communications. With this architecture, the MN is able to handle two mechanism links as defined in Fig. 4.

Mobile node infrastructure: The mobile node is any device capable of communicating with another node. It consists of a notebook or portable computer with WLAN device, GPS receiver, camera and several medical devices that allow the health practitioner to capture a patient's medical data in the form of text, graphics, video, audio and data files, as shown in Fig. 5. Information taken by the peripheral devices is subsequently forwarded over Smart network to health care center or mobile ambulance. The devices are interfaced with the computer through a Universal Serial Bus (USB).

VOICE OVER IP PERFORMANCE EVALUATION OVER MANET

Simulation of VoIP over MANET: The simulation integrates WLAN and ad hoc network to form a wireless-cum-wired environment. The selected routing protocol is AODV and we use an AODV extension called AODV+ (Hamidian, 2003). The performance of AODV+ is evaluated by running simulations on ns2 which is a discrete event simulator developed by the University of

California at Berkeley and the VINT project (Fall and Varadhan, 1997). The AODV+ extensions on ns2 can be obtained (Hamidian, 2007).

In this experiment, a total of 40 nodes with limited mobility, considered as moderate scale network (i.e., 30-100 nodes) (Macker and Corson, 2004) were randomly placed across the simulation environment of size 2500×1000 m. This simulation environment is a model of an emergency situation in an area where 40 emergency units are able to communicate to each other without the needs of communication infrastructures. The units in this kind of environment are mobile but the mobility is limited. Since AODV+ was meant to work in wired-cum-wireless environment, two gateways and a router were deployed to provide connection to the Internet. The gateways and routers formed the wired environment and are within the range of each other. The bandwidth capacity is 1000 Mbps while the delay is 0.1 sec. It is not the purpose of this project to examine the performance in wired network but in the wireless environment. Therefore, the resources in the wired network are assumed to be abundant and shall not affect the overall performance of the system.

The radio propagation range for each node is 250 m and channel capacity is 2 Mbps. The IEEE 802.11 MAC protocol with Distributed Coordination Function (DCF) is adopted as the MAC layer in this simulation model. DCF is a method for nodes to share the channel capacity. The access scheme is Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). This scheme uses Ready-to-Send (RTS) and Clear-to-Send (CTS) control packets to reserve the channel and solve the hidden nodes problems. Each correctly received packet is preceded with acknowledgement (ACK) packet to the sender.

The traffic type is CBR (constant bit rate) which is commonly used for voice based applications (Boukerche and Bononi,2004). In CBR mode, silences are not exploited for other voice or data traffic (Veeraraghavan *et al*., 2001). Each simulation is executed for 200 sec. Data rates are selected from VoIP CODECs and are presented in Table 1, while the summary of the simulation parameters are available in Table 2.

Table 1: VoIP CODECs

CODECs	ITU-T G.711	ITU-T G.723.1	ITU-T G.729A
Packet size (b)	178	82	68
Data rate (kbps)	193	43	108
Packets per second	66	33	100

Table 2 VoIP over MANET simulation parameters

Parameter	Value
No. of nodes	24
No. of gateways	2
No. of router	1
Radio propagation range	250 m
Wireless channel capacity	2 Mbps
Wired channel capacity	1000 Mbps
Medium access control protocol	802.11
Mobility/Movement	Fixed
Simulation time	200 sec
Environment size	1500×1000 m
Traffic type	CBR
No. of connections	4, 8, 12, 16, 20
Queue length	50

Simulation results and analysis: Based on the simulation settings mentioned in previous section, we present the results and analysis in this section. The performance metrics are Packet Delivery Fraction (PDF), normalized routing overhead and end-to-end delay. PDF refers to the ratio of data packets delivered to destination to packets generated by the sources. It affects the maximum number of throughput that the network can support. Normalized routing is where the number of control packets is normalized against sent data packets. It determines the efficiency and scalable of the routing protocol . End-to-end delay measures the time it takes for a packet to reach its destination. It determines how well the protocol uses the available resources efficiently. Figure 6 presents the packet delivery fraction against the various numbers of connections. Based on Table 1 and Table 2, the CODECs introduce tremendous amount of traffic in one second while each nodes are only able to handle 50 packets in their queue. Therefore, we predict that the performance will degrade as the number of connections increase due to limited resources. However, for G.723.1, it is still able to deliver 100% traffic with 4 connections. In the subsequent number of connections, the performance decreased from 84.3 to 40%. G.711 is regarded as the average performer while G.729A is the worst CODEC in this performance metric. As the number of connections increase, more and more packets are being pumped into the network and thus causing cache overflow in busy nodes. G.723.1 is still able to withstand this effect in small connections because the traffic is slower (33 packets sec^{-1}) compare to the others (66 and 100 packets sec^{-1}). Another important factor is the route discovery process. While the originator node spends some time to locate a route to destination, the application layer continues to produce packets. When the queue is full and the route is not yet available, packets on the queue will be discarded.

Average end-to-end metric is presented in Fig. 7. Again G.723.1 is the best performer against the other two CODECs. The delay suffered with 4 and 8 connections is 0.0085 and 0.08 sec, respectively. When it comes to 12 connections, the delay increased tremendously and stabilizes after that. The same situation happened to G.711 during the transition from 4 to 8 connections. G.729A

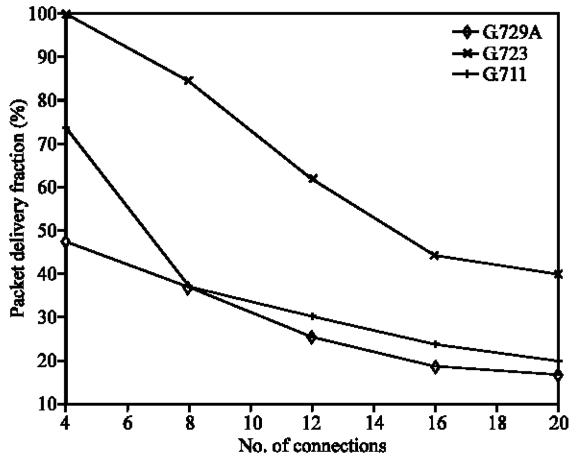

Fig. 6: Packet delivery fraction

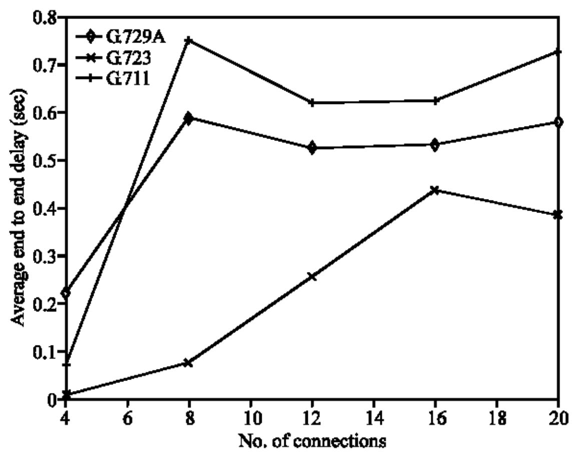

Fig. 7: Average end-to-end delay

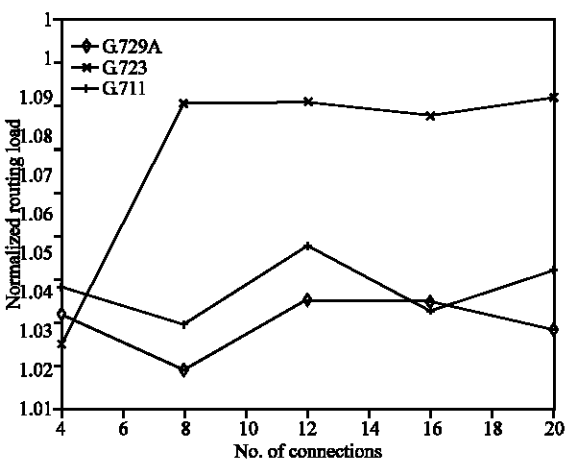

Fig. 8: Normalized routing load

recorded a more stable delay rate in all connections ranging from 0.22 to 0.58 sec. The simulations were done in static environments. The traffics are bound to concentrate on particular routes because change of routes does not occur except transmission error. As a result, the traffic is not evenly distributed and the effect of congestion increases as the number of traffic increases. This is clearly shown in G.723.1 and G.729A. One more interesting factor is the packet size. The larger the size, the more time required to processing them. G. 711 has the largest packet size (178 bytes) followed by G.723.1 (82 bytes) and G.729A (68 bytes). This is the reason G.711 suffered higher delay than G.723.1 except with 4 connections. The relatively low packet size and transfer rate makes G.723.1 the ideal CODEC. Finally, the result for normalized routing load is presented in Fig. 8. This metric is a measure of ratio between routing packets and number of packets sent. The performances for all CODECs are almost similar for all connections. We can conclude that in a static network the amount of routing packets are relative to number of packets sent. Without mobility the nodes do not need to rediscover routes and inform others about broken links.

VIDEO TRANSMISSION OVER MANET

The H.263 (ITU-T Rec. H.263, 1996); Rijkse, 1996), standard by the International Telecommunications Union (ITU) supports video compression (coding) for applications such as video-conferencing and video-telephony. H.263 was based on H.261 CODEC mainly developed for streaming video at bandwidths as low as 20 k to 24 k bit sec^{-1}. H.263 requires only half the bandwidth to get the same video quality as in the H. 261, so H. 263 has largely replaced H. 261. H. 263 uses RTP to transport video streams. The coding algorithm of H. 263 is same used by H. 261 with few improvements and changes to improve the performance and error recovery. Half pixeprecision is used for motion compensation whereas H. 261 uses full pixeprecision and a loop filter. Some parts of the hierarchic structure of the data stream are now optional, so the codec can be configured for a lower data rate or better error recovery. There are four options in H. 263 to improve the performance. They are Unrestricted Motion Vectors, Syntax-based arithmetic coding, Advance prediction and forward and backward frame prediction similar to MPEG called P-B frames. H. 263 supports five resolutions. They are QCIF, CIF, SQCIF, 4CIF and 16CIF. SQCIF is approximately half the resolution of QCIF. 4CIF and 16CIF are4 and 16 times the resolution of CIF respectively. The support of 4CIF and 16CIF means the codec could then compete with other higher bitrate video coding standards such as the MPEG standards. H.263 video streams need to be packetized for transportation over networks. The transport protocol for H.263 streams is the Real Time Transport Protocol (RTP).

Simulation results and analysis: In present experimental setup with parameters as shown in Table 3, we used NS-2 as the simulator for performance evaluation of H.263 video codec over mobile ad hoc network. In telemedicine systems, a videoconferencing session represents the more critical traffic pattern. In this work the videoconferencing sessions had been adopted for evaluating ad hoc routing protocols. The QCIF codification format has been considered as the traffic model for the video conferencing session using H.263 standard, which is termed as low bit rate codification techniques. QCIF implements an image size of 176×144 pixels and a frame rate which differs between 10 and 15 frames per second, generating an average bit rate of around 28.8 kbps (Cherriman and Hanzo, 1996).

Normally QCIF generates a variable bit rate. The combined techniques for dynamic adjustments of codification parameters together with the techniques for buffer management, allow the constant bit rate transmission over a communication channel. Therefore, depending on the type of codification control, a videoconferencing application can generate a constant or variable bit rate. The simulations were carried out several times using a constant and variable bit rate in order to opt for a representative traffic model. The simulations are described as videoconferencing session between two users, which is characterized as a bi-directional and point-to-point connection.

Based on the simulation settings, we present the results and analysis in this section. The performance metrics are Packet Delivery Fraction (PDF), normalized routing overhead and end-to-end delay. PDF refers to the ratio of data packets delivered to destination to packets generated by the sources. It affects the maximum number of throughput that the network can support. Normalized routing is where the number of control packets is normalized against sent data packets. It determines the efficiency and scalability of the routing protocol. End-to-end delay measures the time it takes for a packet to reach its destination. It determines how well the protocol uses the available resources efficiently.

Table 3 Video over MANET simulation parameters.

Parameter	Value
No. of nodes	24
No. of gateways	2
No. of router	1
Radio propagation range	250m
Wireless channel capacity	2 Mbps
Wired channel capacity	1000 Mbps
Medium access control protocol	802.11
Mobility/Movement	Fixed
Simulation time	3600 sec
Environment size	1500×1000 m
Traffic type	CBR
No. of connections	4, 8, 12, 16, 20
Queue length	50
Frame size	10-15 FPS
Bit rate	28.8Kbps 57.6Kbps
Packet size	512 bytes

Figure 9 presents the packet delivery fraction against the various numbers of connections. Based on Table 3 the packetized H.263 video CODEC of 512 bytes introduce tremendous amount of traffic in one second while each nodes are only able to handle 50 packets in their queue. Therefore, we predict that the performance will degrade as the number of connections increase due to limited resources. However, the packetized H.263 codec of 512 bytes, in the low bit rate speed of 28.8 Kbps video transmission environment, it is still able to deliver 100% traffic with 4 connections. In the subsequent number of connections, the performance decreased from 83.3 to 53.7%. In the low bit rate 57.6 kbps high video transmissions the packetized H.263 video code of 512 bytes was able to deliver 99.99% of traffic for 4 connections, for 8 connections it decreased to 86.76% for more increasing number of connections the performance decreased to 66.21 to 38.22%.

Considering the multiple video flow environment, all the mobile nodes are routed to a single gateway and we noted that in low bit rate 28.8 kbps video transmission for 4 connections it was able to produce 100% traffic and for increasing number of connections the performance decreased to 90 to 64% . In the low bit rate 57.6 kbps video transmission, the video codec was able to deliver 100% traffic for 4 connections and for 8 connections it was able to deliver 83% traffic and for more increasing number of connections the performance decreased to 55

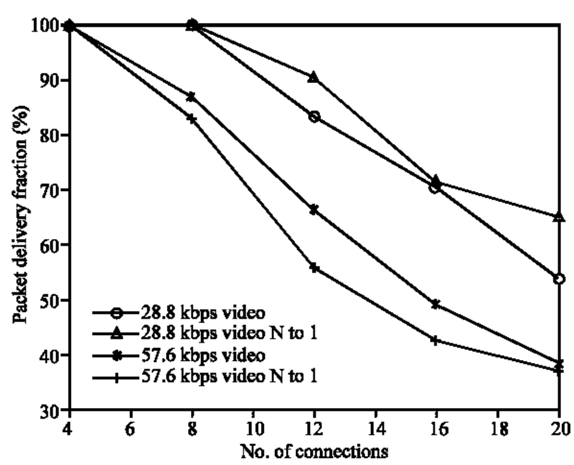

Fig. 9: Packet delivery fraction

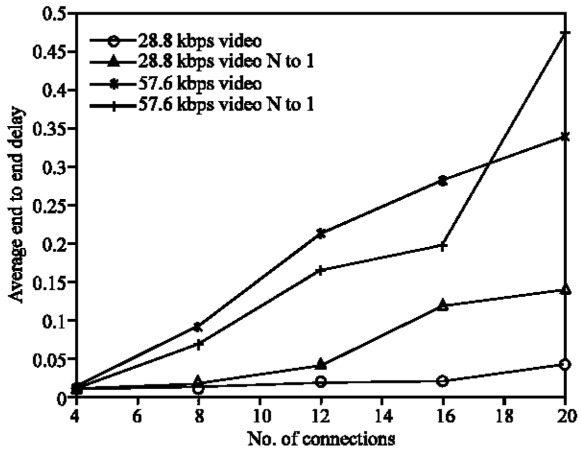

Fig. 10: Average end-to-end delay

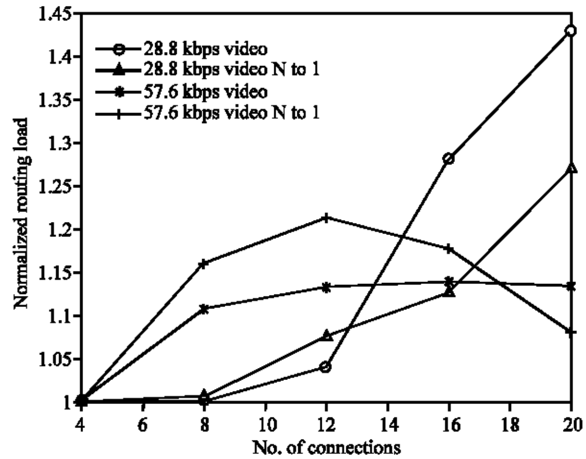

Fig. 11: Normalized routing load

to 38%, due to the buffer caused in the simulation, more packets are being produced and dumped into the gateway causing cache overflow in busy nodes, the heavy load makes the performance decreased, but can satisfy the telemedical needs for video transmission.

Average end-to-end metric is presented in Fig. 10. For low bit rate 28.8 kbps video transmission the delay suffered with 4 and 8 connections is 0.0115 and 0.0125 sec, respectively. When it comes to 12 connections it is 0.0179 sec, the delay increased tremendously and stabilizes after that. The simulations were done in static environments. The traffics are bound to concentrate on particular routes because change of routes does not occur except transmission error. As a result, the traffic is not evenly distributed and the effect of congestion increases as the number of traffic increases. For low bit rate 57.6 kbps video transmission, the delay suffered with 4 and 8 connections is 0.013 and 0.092 sec, respectively. When it comes to 12 connections it is 0.212 seconds and for the increasing number of connections it showed 0.281 to 0.339 sec. For the multiple video simulation, for low bit rate 28.8kbps video transmission the delay suffered was 0.0114 and for 8 connections it was 0.0250 and for increasing number of connections it was 0.040 for 12 connections and increased to 0.140 sec. For the multiple video, low bit rate 57.6 kbps video transmission, the delay suffered was 0.0115 seconds for 4 connections and for increasing number of connections the delay suffered to 0.2 to 0.47 sec. Due to the heavy load on the gateway more packets are delayed in the video traffic, it made the network busy for the high speed 57.6 kbps transmission over MANET.

Finally, the result for normalized routing load is presented in Figure 11. This metric is a measure of ratio between routing packets and number of packets sent. The performance of H.263 packetized video codec of low bit rate 28.8kbps transmission speed for 4 connections is 1.0005 and for increasing number of connections it increased to a high of 1.43. for low bit rate 57.6 kbps high speed transmission for 4 connections it was 1.0005 and later it was stable and same for the increasing number of connections to 1.13. In multiple video flow environment, for low bit rate 28.8kbps video transmission for 4 connections it is noted as 1.0005 and for increasing number of connections it increased to 1.13 to 1.27 for 20 connections. For multiple video of 57.6kbps video transmission for 4 connections it was 1.0005 and for 8 connections it increased to 1.16 and for increasing number of connections it suddenly decreased and stabilized to 1.07, which is a interesting factor for high speed transmission over MANET. It is shown that H.263 codec can perform better for video traffic in high speed transmission for 57 kbps video in both peer to peer and multiple video flow transmissions.

CONCLUSIONS

In this study, the architecture of the mobile tele-emergency system combined with mobileIP, mobileipv6 and the simulations for the VoIP and Video traffic simulation over WLAN based MANET were investigated. In the simulation results, G723.1 worked well in both small scale and medium scale network. H.323 was performed well in high speed transmission over MANET. It can be concluded that mobile tele-emergency system can be implemented in a real time environment in disaster struck areas at a rapid speed without any infrastructure setup.

ACKNOWLEDGMENTS

This research was supported in part by The Ministry of Science Technology and Innovation (MOSTI)

Malaysia, through the Intensification of Research in Priority Areas (IRPA) Programme, under Grant 04-02-10-0041-EA0039.

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