End-to-End Performance of VoIP based on Mobility Pattern over MANETs

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***Abstract*—**In this paper, end-to-end VoIP(Voice over Internet Protocol) performance is evaluated by simulation with NS-2 simulation tool. There are many results studied and published for VoIP performance over TCP/IP networks. But, almost all of them were focused on wired or wireless Internet environments. About MANET (Mobile Ad Hoc Network), VoIP is currently studying several points of research. In this paper, analysis of VoIP performance is done with focusing on the mobility of MANETs. MOS(Mean Opinion Score), network delay, packet loss rates are considered as end-to-end QoS performance parameters.

***Index Terms*—**MANET, MOS, VoIP, Delay, NS-2

I. INTRODUCTION

Nowadays, VoIP called Internet telephony is rapidly supplied and substituted conventional wired telephony. Sooner or later, it will be dominant in the telephone technology and markets.

There are many results studied and published for VoIP performance for wired and wireless(WiFi) TCP/IP networks. The environments for results were the Internet configured with infra-structured networks as general cases.

MANETs are very useful for some special communication environments, like as an emergency, hobby, exploration and military purposes. Their deployment has an expectation be increased gradually. Expanding VoIP to MANETs is very meaningful and useful thing.

MANETs has generally no infra-structured and has to support their own communications and end-devices has multi-function as an end-terminal, relay and others. But this characteristic of MANETs makes some difficult problem like routing, power, relay, QoS, etc.

Research results was published for VoIP on MANETs[1][2][3]. The results were mainly focused on some independent performance parameters such as delay, packet loss rate, etc. MOS(Mean Opinion Score) of VoIP on MANET was studied for some virtual assumed communication environments like as mobility patterns[4].

In this paper, MOS as QoS performance parameter of VoIP traffic is measured for network delay, number of nodes and connections based on various virtual mobility patterns. End-to-end delay, voice packet loss rate and transmission bandwidth of voice packet is also measured for end-to-end performance . Research was done by NS-2 simulation.

This paper is organized as follows. Section 2 presents background. Section 3 describes about the simulation and analyzes performances. Section 4 makes a conclusion.

II. BACKGROUND

A. VOIP

VoIP is a telephony technology operated under Internet as TCP/IP networks using a packet switching protocols. It needs no public switched telephone networks as conventional technology.

Users can make a telephone call if they have a computer and if they can access Internet, without any constraints of location and dedicated terminal.

VoIP will be used extensively and substitute conventional wired telephony rapidly because of it's conveniences and low costs.

But, it needs well-configured networks to handle demands of real-time voice traffic suitably.

B. MANET

MANETs is a network has no infra-structures and easily used for emergency, hobby, exploration and military objects.

Network as a special case, MANETs have some another needs for their smooth operation. That is their own routing, relay, power management, security, etc.

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The traffic pass over them has different characteristics with infra-structured networks. The rate of management traffic is higher the other types of networks.

Making traffic to be low is a significant theme of MANET be studying. Keeping connection between communicating devices under mobility is also one of important issues to be considered.

C. MOBILITY

MANET is basically mobile networks. Generally, all most of all of the devices are moved on their communication areas. Their movement is ceaseless even though there are some pauses to rest.

There are many mobility patterns be studied already. Random Walk, Random Way Point, Manhattan model is typical ones. They are belongs to one category of Random model, Deterministic and Hybrid model.

Random model has arbitrary movement without any constraints. Devices under deterministic model are moved under predefined path. Hybrid model has movement bounded by environmental constraints.

D. E-MODEL

E-model is most widely used scheme as a voice quality evaluation of VoIP[4][5][6].

It was developed by ITU-T as a standard of telephony to estimate voice quality to transfer over it's network.

Output of an E-model is R-factor, obtained from delays, packet loss, impairment factors, and quality call expectations.

R-factor is

$$R = R_o - I_s - I_d - I_s + A \qquad (1)$$

where, R_o is signal-to-noise ratio based on send and receive noise level, I_s accounts impairments that happens with voice signal, example of sidetone and PCM quantizing noise, I_d includes all impairments due to delay and echo effects, I_e means distortion of the speech signal due to encoding and packet loss. Lastly, A is additional factor and represents the degradation in quality used by the users for the case of access. A has a range [0,20] and values of A factor proposed by the ITU-T.

The range of R factor is [0,100], R=0 means an extremely bad quality and R=100 represents a very high quality.

If the value of R factor is computed according to Eq.(1), MOS is easily obtained from R-factor.

For $R < 0$: $MOS = 1$

For $0<R<100$:

$$MOS = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$$

For $R>100$: $MOS=4.5$

MOS, as performance parameter for end-to-end user service quality, has range [1,5]. MOS=0 means extremely bad quality like pure noise, MOS=5 represents perfect fidelity.

ITU-T recommends MOS≥4.0 for conventional wired telephony, MOS≥3.6 for VoIP telephony and cellular telephony.

Other criteria about VoIP performance is network delay. ITU-T recommends that the network delay is below 100ms for a best or high quality. The delay keeps under 400ms even though the system has a poor quality.

III. SIMULATION AND RESULTS

A.SIMULATOR

NS(Network Simulator)2-2.33 is used for this simulation[7]. NS-2 has no ability to handle the VoIP traffic. The VoIP patch, named NS2VoIP, is used for VoIP traffic[8]. NS2VoIP has a function of obtaining the MOS.

Simulator is configured NS-2 ADHOC function and extra mobility files.

Mobility files, called scenario files, are configured with data to describe node mobility. It has initial location, final location, velocity and starting time of movement during simulation run.

These files are generated with scenario file generator which is composed by this study.

B. SIMULATION MODEL

Simulation model is based on mobility patterns. In this paper, six virtual mobility patterns are used to simulate VoIP performance on MANETs, No Movement, Random model, Directional model, Group model, Assembly model and Dismissal model. First three models were the models which were used at [4]. Other three models are newly adopted in this paper.

First three models works as follows written in [4]. In No Movement model, all of MANET node dose not moved during entire simulation period. In Random model, all nodes are moved according random position with random velocity. The velocity is below 2m/s as a human walking velocity. 2m/s mean 7.2km/h movements. In Directional model, all node has movement to same direction from their initial location with predefined random velocity same as Random model.

Newly suggested three models work as bellows. Group Models has group movement of nodes. Each nodes move to their destination. Destination is different from each other as their group. In this paper, five groups are used and five locations are used to give destinations for each group. One location for a group is center point of network range. The location of center point will be (335, 335) in 670X670 network size. Other four locations for groups are assumed as vertices of network square. Number of nodes belongs to each group is varied on number of nodes of MANET. For 20 node MANET, number of nodes to each group is (V1,V2,V3,V4,C)=(2,2,2,2,12). Vn means Vertex n point and C means Center point. 30 nodes MANET has (4,4,4,4,10), 40 nodes MANET has (7,7,7,7,12) and 50 nodes MANET has (10,10,10,10,10).

In Assembly model, all nodes of MANET move to a specified Assembly location, center point in this study. Dismissal model has opposite movement of Assembly model. All nodes are go move away from their initial point to nearest side area of MANET.

Data for mobility patterns has obtained using pseudo random number generator. In No Movement Model, data is only initial location. For Random model, the data for final destination of movement and the velocity is appended to the No Movement model. So, the initial location of each model is the same. Finally, in Directional model, one of the destination location(x,y) is selected as same value for all nodes. So, all nodes will be moved to same direction. Each node has same velocity with Random model. Mobility pattern of Group model, Assembly model and Dismissal Model are manually configured with predefined locations suitable for their definitions.

The mobility pattern about each mobility model for each simulation condition is generated with scenario file generator described former section, A.Simulator.

Other basic important parameter used in the simulation is

Routing : DSDV
MAC : 802.11
Communication Area : 670×670
Node : 20/30/40/50.

Some important VoIP parameters used in the simulation is

Network delay : exponential distribution
Codec : G.711
Packet Loss Rate : 0.0

But, some packets may be lost according network situation when they transferred over network.

C. RESULTS AND ANSLYSIS

In this section, performance results of simulation for MOS, end-to-end delay, packet loss rate and transmission bandwidth are presented according to number of connection, number of nodes and network delay. In all cases, simulation was executed for 300s.

Figure 1 show measured MOS as a function of number of VoIP connections. In figure 1, number of nodes is 30, mean network delay is 1ms.

Assembly model has best performance with small number of VoIP connection. Group model has better performance with increase of VoIP connection. But all mobility models except No movement has a similar performance. It means that movement makes a better chance to connect each node, unrelated their patterns.

The results also mean that connection may keep under 6 for normal use(MOS≥3.6) and 10 for well trained usage(MOS≥2.5) to having recommended voice quality for all of mobility pattern.

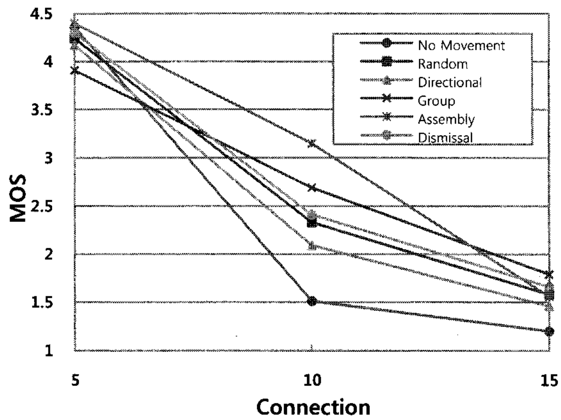

Fig.1 MOS as a function of number of connection.

Figure 2 presents MOS versus network delay. Under 200ms, MOS keeps good results. The results were obtained from 30 nodes, 5 VoIP connections. All models have good results for network delay.

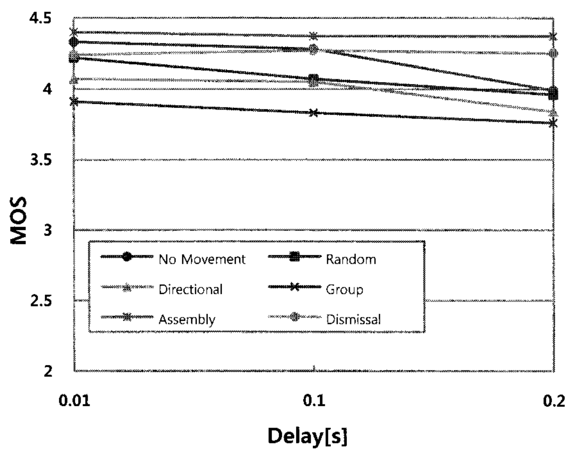

Fig.2 MOS as a function of network delay.

Figure 3 describes MOS as a function of number of nodes. The results are collected from 5 VoIP connections, 10ms network delay. Under 40 nodes, 5 VoIP connections has good MOS. For 50 nodes, Assembly, Dismissal and Directional Model has good MOS, but No Movement model has very low MOS, Random model and group model has a poor MOS.

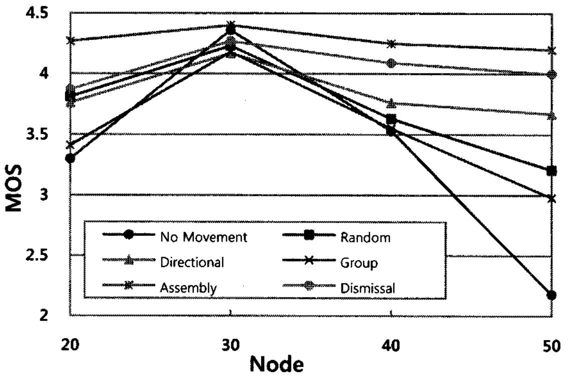

Fig.3 MOS as a function of number of node.

Figure 4 show end-to-end delay as a function of number of connection. The results are collected for 30 nodes. Under 5 connections, network keeps under 100ms as a ITU-T recommendation for a best/high service quality. But, with increase of connection, end-to-end delay is very highly increased.

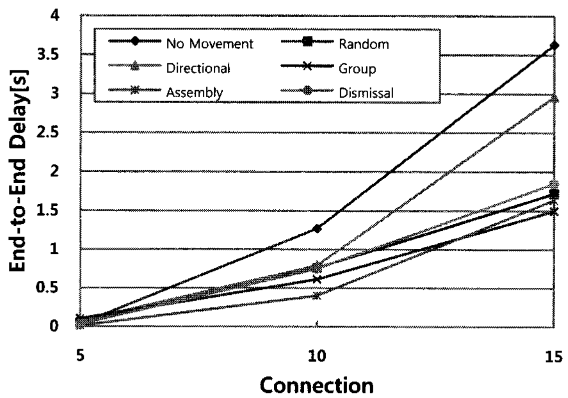

Fig.4 End-to-End delay as a function of number of connection.

Figure 5 describe packet loss rate as a function of number of connection for 30 nodes MANET. In figure 5, packet loss rate of Assembly mobility is best then other mobility models. Random model, Directional model, Group model and Dismissal model has good results. No Movement model has a poor performance.

Figure 6 shows bandwidth of each mobility models for a MANET which has 30 nodes. Bandwidth is obtained from (VoIP packet size) X (VoIP Packets sent per second). VoIP packet has 40 Bytes header and variable VoIP payload depending on codec. For G.711 VoIP payload is 80 Bytes. Bandwidth is 6~7Mbps for 5 connections, 10~14Mbps for 10 connections and 12~19Mbps for 15 connections. It means that each connection has approximately 1.2Mbps bandwidth.

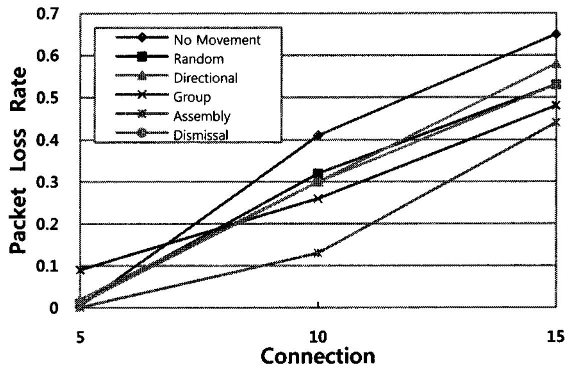

Fig. 5 Packet loss rate as a function of number of connection.

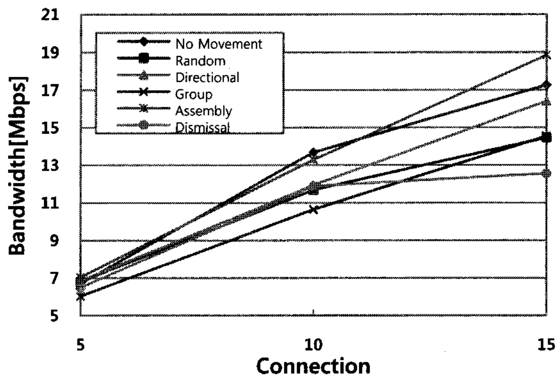

Fig. 6 Bandwidth as a function of number of connection

Figures show that the voice quality of VoIP on MANETs is almost good for all movement models except No Movement.

IV. CONCLUSIONS

In this paper, MOS, end-to-end delay, packet loss rate and bandwidth as end-to-end VoIP performances of MANETs based on mobility patterns are evaluated. NS-2 simulation package with NS2VoIP patch is used to measure the parameter.

Six mobility patterns, No Movement, Random, Directional, Group, Assembly and Dismissal model are used for simulations.

All models exececpt No Movement has good VoIP performance.

Through the simulation, appropriate number of VoIP connection is checked below 6 for normal use and 10 for a well-trained person.

There are needs to more detailed simulation and analysis to analyze this number. And solutions to improve the number of connection supporting MOS condition are also needed.

Performance evaluation of VoIP on MANET for real mobility pattern is also needed.

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