

ACIP over MANET: OLSR for Remote Broadcasting

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Abstract: For the means of remote broadcasting in radio broadcasting, Internet Protocol (IP) technology has been started to consider. We have been studying the feasibility of introducing audio over IP over mobile ad hoc network for the future remote broadcasting. In this study, we investigate the performance of mobile ad-hoc network protocol, OLSR for audio contribution over IP using mobile phones as intermediate nodes. We used ITU G.711 as the outgoing audio codec. Through a simulation study by the use of a network simulator, the following quality of service parameters were assessed: packet loss, end-to-end delay, jitter and Mean Opinion Score (MOS).

Key words: ACIP, AoIP, OLSR, remote broadcasting, MANET

INTRODUCTION

Radio broadcasting is old but reliable media. For example in Nigeria, most people (87.4%) in the urban and rural areas listen to the radio to get information daily and 39.4% use their mobile phones with inbuilt FM receivers to listen to the radio (ITU-T, 2016). Mobile phone users in Nigeria has grown astronomically since 2001 when mobile wireless phones (GSM) was introduced in Nigeria. Presently, >50%, 90 million of the population in Nigeria own smartphones (NCC, 2017).

On-site remote (live) broadcasting is indispensable in radio broadcasting. It had been achieved by use of analog public telephone lines or analog private radio channels. Analog public telephone line is gradually replaced to ISDN and it has been used for remote broadcasting. However, due to recent explosive popularization of the internet, it is expected that these ISDN and analog radio channels will be replaced to internet channels from the viewpoint of costs and performances. Actually, Audio Contribution over Internet Protocol (ACIP) is increasingly used by radio broadcast companies (Graubner *et al.*, 2010). ACIP is based on Audio over IP (AoIP) which is the distribution of digital audio via IP.

Audio Contribution over Internet Protocol (ACIP) is increasingly used by radio broadcast companies as ISDN are gradually being phased out. ACIP is based on Audio over IP (AoIP) which is the distribution of digital audio via IP (Graubner *et al.*, 2010). In Federal Radio Corporation of Nigeria, ISDN is commonly used as Point to Point (PTP) links between remote locations and main studios. ACIP is rather novel over ISDN and not yet used (Oyakhire and Gyoda, 2016).

There are two major types of audio contribution in broadcasting: unidirectional and bidirectional audio. Usually, bidirectional audio is used in remote broadcasting where the return audio is narrowband which is used for cueing the contribution.

Transmission of multimedia traffic over Mobile Ad-hoc Networks (MANETs) is quite a challenge due to several constraints which include: high mobility of the nodes, frequent changing topology, hard delay, etc. These constraints make the transmission of multimedia traffic over such networks a challenging task. Also, delivering audio via wireless networks can be challenging because excessive delays or loss of packets may cause interruptions in the audio. We have been studying feasibility of introducing AoIP over MANET for the remote broadcasting (Oyakhire and Gyoda, 2016).

In this study, we focus on ACIP suitable for radio broadcast over MANET. Explicitly, the performance of audio over MANET using PCM G.711 codec and OLSR routing protocol will be studied. Riverbed modeler 18.0 will be used to run several simulations. The following quality of service parameters will be assessed: packet loss, end-to end delay, jitter and Mean Opinion Score (MOS).

MATERIALS AND METHODS

ACIP components

OLSR: For this study, we use a proactive routing protocol, Optimized Link State Routing (OLSR). Due to its proactive nature, it has the advantage of having the routes available when needed. Instead of using all the



Fig. 1: Simulation model

nodes in a MANET, it declares a subset of nodes with its neighbours that are its multipoint relay selectors and minimizes the flooding of control traffic by using only the selected nodes called the Multi Point Relay (MPR). This technique reduces the number of retransmission in a broadcast procedure (Oyakhire and Gyoda, 2016).

Audio codecs: For audio contribution over IP, there are mandatory codecs stipulated by EBU as they are cheap and less complex to implement, they include ITU G.711, ITU G.722, MPEG layer II and PCM (EBU-TECH, 2008, 2014). For this study, we implemented ITU G.711 codec as the audio codec at the Source node. G.711 codec is a public domain codec and popularly used in almost all Voice over IP systems. Its bit rate is 64 kbps and a frame size of 20 msec per RTP packet.

Also, G.729A CS-ACELP is used at the destination node for cuing the contribution. Its bandwidth is low and a bit rate of 8 kbps. High-quality audio cannot be transported with this codec, it is just used for voice calls only.

Quality of service parameters for ACIP: As ISDN for remote broadcasting is set to be phased out in the future and replaced by packet switched technology, the quality of service parameters change as well. ISDN is set to be phased out in the future being replaced by packet switched Voice over IP (VoIP) technology run on managed networks as well as on the best effort internet which both provide no more than statistical guarantees. In contrast, the ISDN QoS satisfies the broadcaster’s audio contribution needs by providing guaranteed services meeting the requirements with high availability. In VoIP networks, these guarantees disappear and QoS parameters such as data rate and network delay vary over time.

The QoS for ACIP is measured by performance metrics such as packet loss, end-to-end delay, jitter and Mean Opinion Score (MOS). Packet loss is measured as a percentage of the packets lost to the packets sent. End-to-end delay is measured as the delay from the Source to the destination node. It includes network delay, audio encoding and decoding delays and compression and decompression.

Table 1: QoS requirements for ACIP

| QoS parameters | ACIP |
|------------------|-----------|
| Packet loss | <<0.2% |
| End-to-end delay | 100 msec |
| Jitter | <<40 msec |
| MOS | 4.1 |

Table 2: Simulation parameters

| Simulation parameters | Values |
|---|---------|
| Data rate (mins) | 65 Mbps |
| Distance between source and destination | 500 m |
| Distance between intermediate terminals | 45 m |
| Number of intermediate nodes | 10 |
| Node transmission range | 50 m |
| Simulation time | 10 min |
| Transmit power of intermediate nodes | 0.1 W |
| Transmit power of intermediate nodes | 0.2 W |
| Mobility model | Static |

Jitter is the variation in arrival time of consecutive packets. Mean opinion score scale varies from 1-5 and it measures the quality of audio. The worst quality is 1 and the best quality is 5. The requirements for QoS parameters for ACIP provided by the EBU/N/ACIP group is summarized in Table 1 (Graubner *et al.*, 2010).

ACIP simulation and parameters: To evaluate QoS of ACIP on the MANET, Riverbed Modeler 18.0 was used for the simulation. The simulation models for ACIP at the remote end is shown in Fig. 1. The ACIP application is defined as a voice application running streaming multimedia service and the audio frame packets are generated discretely.

For this simulation, we assumed the location is outdoors thus the intermediate terminals has a wider transmission range than indoors. We proposed a transmission range of 50 m (150 feet) for the intermediate terminals. All terminals as well as the source and destination nodes are 45 m apart thus comfortably in the range of transmission of neighboring nodes. The distance from the source node to the destination node is 500 m. Other parameters for the simulation are given in Table 2.

RESULTS AND DISCUSSION

For remote broadcasting, there are a number of quality of service parameters that can verify the quality of

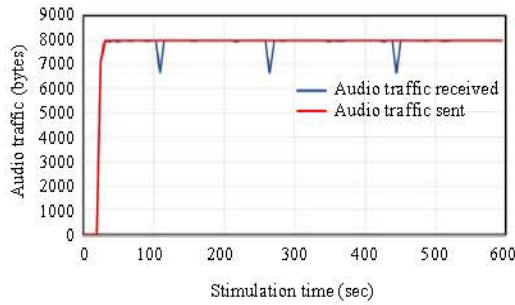


Fig. 2: Packet loss

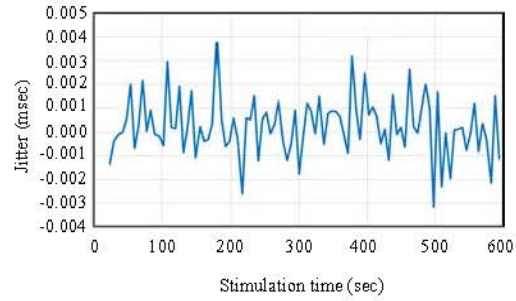


Fig. 4: Jitter

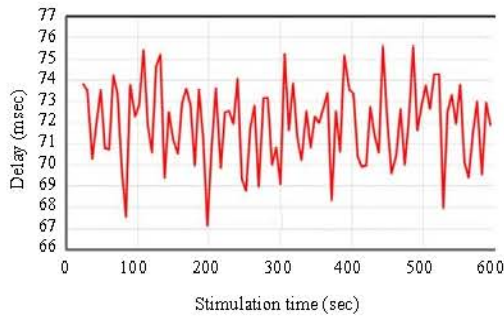


Fig. 3: End-to-end delay

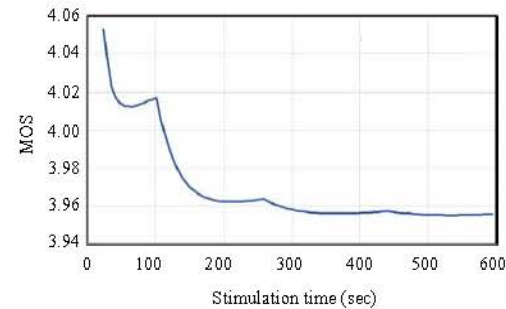


Fig. 5: Average MOS

the broadcast. For this study, the QoS parameters that will be discussed are: packet loss, packet end-to-end delay, jitter and Mean Opinion Score (MOS). Ideally, MOS is a Quality of Experience (QoE) parameter but in this simulation, MOS is estimated virtually by riverbed modeler.

Packet loss: For this simulation, 8000 bytes/sec of audio packets is sent. In remote broadcasting, it is important to ensure that 100% of the packets are received at the destination. However, this may not hold true. Simulation result in time change of sent and received at the destination. However, this may not hold true. Simulation result in time change of sent and received packets is shown in Fig. 2. During the simulation, there are three periods where the audio packets received is about 83% of the audio packets sent. In total, the packet loss is 0.59% this is $>0.2\%$ thus, not satisfying the ACIP requirement.

End-to-end delay: The end-to-end delay is taken for every audio packet during the time of the simulation. The simulation result in time change of end-to-end delay is shown in Fig. 3. The delay for ranges from 67.2-75.6 msec. This satisfies the ACIP requirement (<100 msec).

Jitter: Simulation result in time change of the jitter is shown in Fig. 4. The jitter varies from -0.0031 to 0.00374 msec this is rather negligible as the simulation result satisfies the ACIP requirement (<40 msec).

Mean opinion score: The MOS for this simulation varies peaks at 4.05 at the start of the simulation and drops to a value of 3.95 towards the end of the simulation. The MOS value does not satisfy the ACIP requirement (Fig. 5).

CONCLUSION

In this study, we simulated ACIP for remote broadcasting using static mobile phones (iPhones) as the intermediate terminals. The results for this simulation conforms to the ACIP QoS requirements except for the packet loss. This may be due to the high number of intermediate nodes.

SUGGESTIONS

In the future, we will use raspberry Pi and iPhones for real simulation of remote broadcasting. The quality of experience will be measured by people subjectively.

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