A Study on AoIP over MANET for Remote Broadcasting

Omuwa OYAKHIRE and Koichi GYODA
Graduate School of Engineering and Science, Shibaura Institute of Technology
3-7-5 Toyosu, Koto-ku, Tokyo 135-8548, Japan
E-mail: omoyakhire@gmail.com, gyoda@shibaura-it.ac.jp

Abstract: Audio over Internet Protocol (AoIP) is one of the emerging protocols that is used for a means of streaming high-quality audio feeds over IP Networks from a remote site to the main studio in radio broadcasting. The AoIP in this study is proposed to be carried over a continuously self-configuring infrastructure-less network of mobile devices refered to as Mobile ad-hoc Networks (MANETs). This study will focus explicitly on the performance of AoIP over MANET with different codecs and proactive protocols. Throughout a simulation study by use of network simulator, the following Quality of Service parameters will be assessed: throughput, delay and packet loss to evaluate the uniqueness of this system.

Keywords-- **AoIP, MANET, OLSR, QoS**

1. Introduction

Audio Contribution over Internet Protocol, also known as Audio over Internet Protocol (AoIP) is being used by radio broadcasting companies for streaming of high-quality audio feeds over IP networks from a remote sites or local offices into main studio centers [1]. The IP networks used can be private networks with controlled Quality of Service or the Internet, which is used over long distances. AoIP has become increasingly used by radio broadcast companies as several countries are withdrawing ISDN services which was heavily used for audio contribution in the past [2].

There are two major types of audio contribution in broadcasting: unidirectional and bidirectional audio. Also, two different types of audio transfers can be used over IP networks: audio transfer and live streaming. This study will focus on live streaming.

There is a significant difference between VoIP and AoIP. In VoIP, high signal compression, undisturbed data flow across network boundaries (routability) and universal / common connection protocols are key and signal quality and latency are less important. On the other hand, professional audio / broadcasting application, full signal transparency, very low latency and a high channel count, i.e. number of available channels of audio used for broadcast, are the decisive factors in AoIP.

AoIP is the protocol that distributes audio services (voice, background music, foreground music etc) across the internet of next generation networks (NGN). The goals and quality parmeters for AoIP are summarised in Table 1. Another major difference is the requirement for unidirectional one-to-many connections in the professional audio / broadcast field, whereas with VoIP most connections are essentially point-to-point bidirectional.[4]

The advent of wireless networks in Nigeria is as recent as 2001, as the introduction of GSM services in Nigeria broke the monopoly of the government owned telecommunication services. In the space of fifteen years, the wireless services grew astronomically. In Nigeria, there are four major mobile network operators that provide wireless telephone services as well as internet services with over ninety million (90,000,000) people having access to wireless services[5].

Table 1. QoS requirements: AoIP vs VoIP[3]

QoS Parameter	**AoIP**	**VoIP**
End to end delay	<100 ms	150-400ms
Audio Bandwidth	12-20kHz	3.5-7kHz
Packet loss	<< 0.2%	<0.2%
Data rate	80-2000kbit/s	20-80kbit/s
Audio Quality	MPEG/Eapt-X	G.7XX/AMR

In the Federal Radio Corporation of Nigeria, AoIP is rather novel as point to point (PTP) studio-transmitter links over ISDN, and is commonly used for Remote Broadcast from remote locations to the main studio. The major challenges using this means are latency, audio quality and restricted distance coverage.

Recently, Mobile Ad hoc networks (MANETs) laid foundation for a wide range of applications like emergency services, wireless sensor networks and vehicular networks. However, several constraints exist in MANET's such as high mobility of the nodes, frequent changing topology, hard delay, etc.[5] These constraints make the transmission of multimedia traffic over such networks a challenging task. Routing protocols are an important issue in mobile ad hoc networks since there is no central manager entity in charge of finding the routing path among the nodes. Using MANET for remote broadcasting can offer a less expensive solution as well as its easy implementation. However, delivering audio via packet switched networks can be challenging because excessive delays or loss of packets may cause interruptions in the audio.

This study will focus on AoIP transmission suitable for radio broadcast over MANET. Explicitly, the performance of audio over MANET using different codecs and routing protocols will be studied. Network Simulator will be used to evaulate the Quality of Service (QoS) such as throughput, delay and packet loss.

2. Proposed MANET Protocol

For this study, we propose 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 links in a MANET, it declares a subset of links with its neighbours that are its multipoint relay selectors and minimizes the flooding of

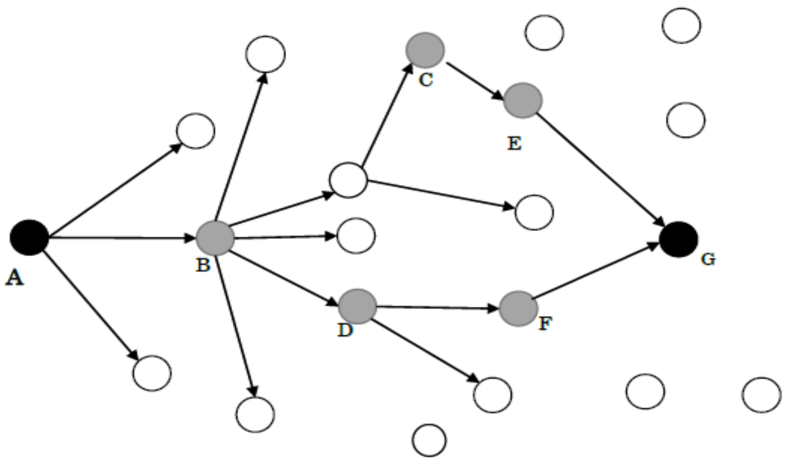

Figure 1. Depiction of OLSR with its MPR

control traffic by using only the selected nodes called the multipoint relay (MPR). This technique reduces the number of retransmission in a broadcast procedure[6].

Each node in the network selects a set of MPR nodes in its neighborhood which retransmits its packets of that node. The neighbours of a node which are not in its MPR set, read and process the packet but do not retransmit the broadcast packet received from the node. Each node selects its MPR set among its one hop neighbors such that the set covers (in term of radio range) all the nodes that are two hops away. Every node in the two hop neighborhood of a node must have a bidirectional link towards its MPR. The smaller the MPR set, the more optimal is the routing protocol.

Each node must detect the neighbor nodes with which it had a direct and bi-directional link. To do this, each node periodically broadcast HELLO messages, containing the information about its neighbors and their link status. These control messages are transmitted in the broadcast mode and are received by all one-hop neighbours but not relayed to further nodes.

In Figure 1, A represents the source and G is the destination. Nodes B, C, D, E and F are the MPRs.

3. Proposed Audio Codecs

An audio codec is a term used for conversion of analogue audio signals to digital signals for transmission and from digital to analogue signals. There are many codecs available for audio available for AoIP. The most common codecs are the G.7XX series that are used for VoIP applications. In this study, we evaluated two codecs, ITU-T standards G.711 and G.722.

G.711 PCM codec is a popular codec used in VoIP. The audio bitrate is 64kbps and a sample rate of 8 kHz. It is also used by all Public Switched Telephone Network (PSTN) and ISDN. Default frame size is 20ms of audio per RTP packet.

G.722 ADPCM codec is a standard ITU codec that provides a sample rate of 7 kHz at different audio bitrates ranging from 48-64kbps. Default frame size is 20ms of audio per packet, which is the same with G.711. It is easy to implement despite a limited audio bandwidth.

Table 2. Audio codecs considered for simulation

Audio Codecs	**G.711**	**G.722**
Coding techniques	PCM	ADPCM
Audio bitrate (kbit/s)	64	64
Sampling rate(kHz)	8	7
Frame size (milliseconds)	20	20
Compression and decompression delay (secs)	0.001	0.001

Table 2 summarizes the characteristics of the audio codecs G.711 and G.722.

4. Proposed QoS Parameters

There are many parameters that can verify the quality of an AoIP. The Quality of Service (QoS) parameters used in our study are throughput, end-to-end delay and packet loss as these parameters are very important in remote broadcasting.

Throughput: This is the maximum number of bits that can be sent through the channel per second. It is directly related to the low layer (wireless LAN) specification, including wireless frequency band and modulation method.

End to end delay: This is the length of time taken for a packet to travel from its source to reach its destination. For AoIP, the end-to-end delay as stipulated by ITU should be less than 100ms.

Packet Loss: Audio over IP is susceptible to packet loss. Due to the sensitivity of AoIP, packet loss should be at a minimum of 0.2%. Packet loss is measured as the percent of packets dropped at the receiver prior data stream playback.

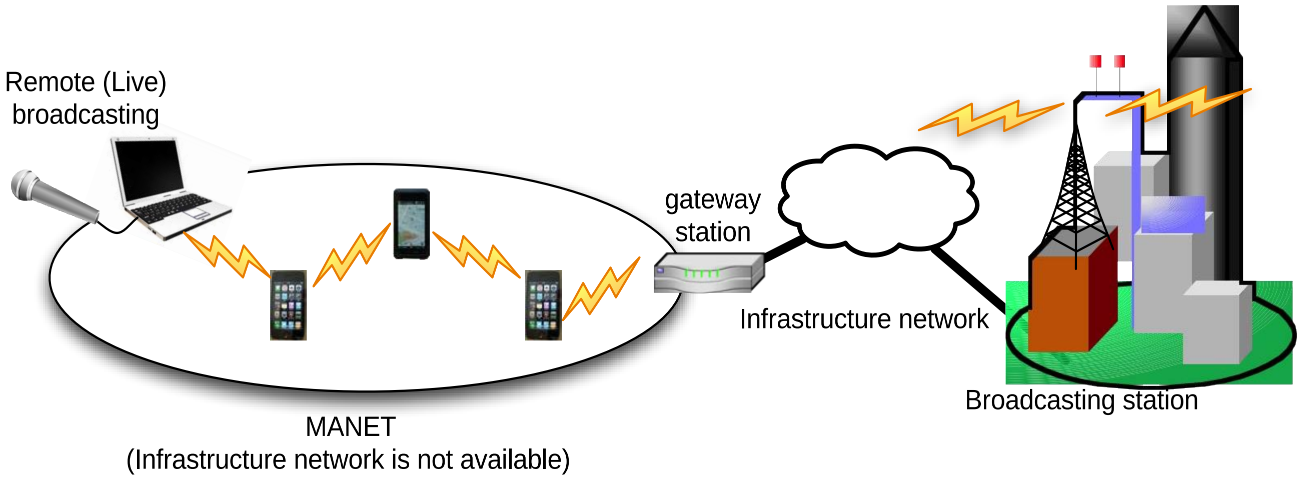

Figure 2. Network model to evaluate proposed protocol.

Table 3. MANET simulation parameters

PARAMETERS	Values
MANET protocol	OLSR
Physical and MAC layer	IEEE 802.11n
Frequency band	5GHz
Data rate	65Mbps
Node transmission power	0.1W
Node transmission range	From 100 metres to 200 metres
Area Dimension	1km x1km
Number of Nodes	50
Mobility models	Static and Random
Simulation Duration	10 minutes

5. Proposed Simulation Parameters

5. 1 MANET Simulation Parameters

Figure 2 depicts the network model to evaluate the proposed protocol. In this study, simulation of AoIP over MANET will be performed by use of a network simulator. We set MANET simulation parameters as shown in Table 3.

We propose to use the IEEE 802.11n wireless LAN as the physical and MAC layer for the OLSR protocol. Frequency band and data rate of IEEE802.11n is set to 5GHz and 65Mbps, respectively. Transmission power of each node is 0.1W, and transmission range is set to 100m to 200m.

MANET area where the infrastructure network is not available is assumed to one square kilometers. In this area, we set 50 nodes, though the number of nodes depends on the population density of having mobile nodes, e.g. mobile phones. The simulation will be applied over two types of mobility models: static and random. In the static model, the MANET nodes are stable and not moving while the MANET nodes are moving in random directions in the random model. Simulation time will be set to 10minutes.

5. 2 Application Configuration

In this paper, the voice application is selected for this simulation. A new application will be created and called “AoIP”. This application will be used for all scenarios with different audio codecs.

5. 3 Profile Configuration

A unique profile called Audio Profile will be created. Its profile settings and configuration will be created based on the audio codecs and the application, AoIP created.

6. Conclusion

As AoIP is gradually becoming the means of remote broadcasting in radio broadcasting and MANETs an emerging field in networking, this paper aims at using MANET for remote broadcasting. The use of MANETs in remote broadcasting is an innovative field. We proposed a MANET protocol, OLSR and used two audio codecs, G.711 and G.722 for the simulation. The Quality of Services parameters: throughput, delay and packet loss will be compared for the two audio codecs for best performance.

In future work, we intend to simulate AoIP over MANET with other audio codecs. We will also evaluate the quality of the audio subjectively using suitable test techniques.

References

[1] EBU Doc TECH 3329, "A tutorial on Audio Contribution over IP," Geneva, 2008.

[2] EBU Doc TECH 3326, "Audio Contribution over IP-Requirements for Interoperability, " Geneva, 2014.

[3] M.Graubner et al, "A new QoE model and evaulation method for broadcast audio contribution over IP," *Proceedings of NOSSDAV2010*, pp.57-62, June 2010.

[4] Introduction – Why Audio over IP? , http://ravenna.alcnetworx.com/technology/why-audio-over-ip.html

[5] Nigerian Communication Commission–Industry Information- Industry Overview, http://www.ncc.gov.ng/index.php?option=com_content&view=article&id=68

[6] Said El Barak et al, "VoIP over MANET (VoMAN): QoS & Perfromance Analysis of Routing Protocols for different audio codecs," *International Journal of Computer Applications*, vol. 36, no.12, December 2011.

[7] T. Clausen and P. Jacquet, "Optimized Link State Routing Protocol," IETF RFC3626, https://www.ietf.org/rfc/rfc3626.txt