Doctoral Thesis

博士論文

Enhanced dependability and feasibility in multi-hop relay networks considering finite buffer and power in relay nodes

（ 和文訳：中継ノードの有限記憶と電力を考慮したマルチホッ プリレーネットワークにおける際立つ高信頼性と実現性）

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Abstract

Network delay or latency is targeted to be one of the bench marks in 5G and beyond networks. These networks are structured as having smaller cells and therefor to have more coverage need dense deployments. As such new paradigms of network design need to be considered from the onset. In low power networks like wireless sensor, most of the traditional protocols do not hold because of the resource constrain in power and memory and processing capabilities. The purpose of this work is to study the impact of finite buffers in the overall delay performance of packets in a heterogeneous traffic environment. In heterogeneous traffic environments different classes of traffic have different quality of service requirements. With this in mind, we propose and design a virtual buffer extension or a distributed buffer storage algorithm for nodes in a network depending on the topology. The topology considered in this work is multihop mesh network structure and also networks with multiple routes. The structure is as follows the main base stations are arranged in a multihop fashion which act as intermediate relays. Client or mesh nodes cluster around each basestation ,this results in the base station carrying both its mesh clients traffic and that from the other basestation in the same hierarchy. We begin by building a case on the theoretical aspects of buffer extension for consideration to use in practical systems. the ultimate objective is to study different topologies in a mesh fashion and the interplay with network coding and routing.

Buffer expansion may find application in resource constrained environments like wireless sensor networks (WSN) and Wireless Body area networks (WBANS) as well as application in low power wide area networks such as sigfox and LORA. This also has can have application in developing nations where the digital divide is large and the use of low cost networks in such environment can deliver the needed infrastructure for medical and social services use. As future work, we would like to study the effects of implementing such a technique on the power consumption of a node and network lifetime.

あらまし

日本語のアブスト

本研究の主課題は、５G、IoT/M2M などの近年の情報通信ネットワークにおける信頼性、安全性などの dependability に注目し、実現性を度外視してネットワークを理想化した理論的な最適を求める従来のアプローチに対して、センサーネットワークや無線ボディエリアネットワークなどのアドホックネットワークを中心とするマルチホップリレーネットワークにおける中継ノードの有限サイズのバッファや電池残量などの現実の制約を考慮した実用性の高いネットワークのルーティングやその発展としてのネットワークコーディングを考案し、性能解析することにある。その具体的な提案として、まず、ネットワークのルーティングにおいて、主となる中継ノードのバッファサイズを補うために、その周辺ノードにより不足するバッファを導入し、パケットロスを改善すると共に、ネットワークのトラヒックに柔軟に対応しその結果としてスループットの向上を得る。次に、ノード毎のバッファサイズだけでなく電池残量も考慮し、災害時のネットワーククラウドを高信頼に長時間利用できるネットワーク寿命を延ばすことにつなげる。さらに、ルーティングにより求められたソースとシンクを繋ぐ複数のルートを用いて、高信頼性を確保できるネットワークコーディングを考案し、実用環境に適した性能を得る。以上より、必ずしも高価な高信頼ノードが得られない途上国のネットワークインフラストラクチャにおいても、高信頼なネットワーク性能を実現できることを示す。

Chapter 1

Introduction

1.1 Background

1.2 Motivation and Aim

The motivation behind this thesis is two fold:Social Aspects and technical aspect. The cost of communication infrastructure and access to communication facilities is still a major challenge in developing countries. With the available infrastructure,the cost of access is record high.Many nations have rolled out agendas and policies towards bridging the digital divide which has seen introduction of Information and Technology (ICT) training in schools to help both students and teachers in accessing information. The result has also been the digital migration of terrestrial television from analog systems to the now digital systems. Inspite of all these efforts,the cost of communication infrastructure and access still remains high for the avarage individuals especially in rural areas.

The advent of coming of mobile communication in developing countries especially in Africa has revolutionised the way business is done, case in point the success story of MPESA mobile money which has been a great success and has been replicated in various countries world wide. With great strides in one area,other areas of importance remain undeveloped for instance Health care industry as well as eduacation especially in the rural areas. At the heart of all these issues is the need for dependable and cost effective infrastructure to provide communication services.

With Technology Motivation being the other motivating,given the backdrop of the social issues as outlined previously,there has been a lot of technologies which have been deployed, some with very little success post project implementation and some have been successful. As mentioned most rollouts come at huge costs and capital investment and present a high operation cost (OPEX).Most ventures are usually private entities ,for profit organisations and so since most rural areas have a low average revenue per user (ARPU),most businesses not invest in such areas.

However,there has been a steady increase in a number of technology which are currently on the market and can help fill in some of these divides.The coming in of wireless sensor networks,low power wide area network technology provide a good promise.One characteristics is them being low cost ,however they are limited in terms of resourses,in other words resource constrained.Most have low memory ,low battery,low processing speeds and low data rates to mention a but a few.despite these,they still can play a critical role in brigding the digital divide.To deploy and redesign these networks is one of the ways in which this can be met.these entails studying new network topologies ,new protocol design and how all these technologies can coexist together without affecting the operation of the other.This provides the motivation of this thesis and aims at highlighting some of the aspects thats can be redsiigned to fit in the future paradigm of communication networks.

1.3 Problem Statement and Assumptions

Villages in developing countries are arranged in clusters spread over vast geographical areas. Multi-hops networks interconnecting these clusters is one of the architectures that can provide digital services by using wireless access points.To avoid congestion many deployments have a maximum number of users that the access point can accommodate.The other aspects of the network that are looked at are that the network will contain multiple routes from source to destination depending on the node density.With a high node density,the nodes are closer together and therefore easy to establish multiple routes.

In order to provide such access at an affordable cost, low networks deployed would have to carry a multiplicity of services like medical application, money transfer services as well as mobile money. Mesh networks and Multi-hop become ideal candidates for these kinds of deployments. If the available Low Power Wide Area Networks (LPWAN) could be deployed in Multi-hop fashion, within each village cluster and network access fulfiled.

This thesis proposes a Virtual Buffer Extension of finite buffers in intermediate nodes of Multi-hop relay networks.These buffers form a distributed storage system such that instead of packets being lost ,they are transferred and stored in the neighbouring node's buffer.The second contribution is the design of a network coding protocol and performance is evaluated combined with virtual buffer extension on packet delay.No mobility in the network is assumed and the network topology remains fairly constant during evaluation.

1.4 Thesis Objective and Contributions

In order to achieve enhanced dependablity,the feasibility of two techniques are investigated.Fistly aVirtual Buffer for relay nodes to aleviate congestion and avoid Packet delay.Secondly implementation of a Network code as a protocol in the application and /or network layer to improve success probability in multi-routes in multi hop networks and when packets are tranfered to the sorrounding nodes in virtual buffer extension.The contribution thus achieved is implementation of a network coding module and its feasibility for use in Virtual Buffer Extension.

1.5 Related Technologies and Topics

1.5.1 State of 5G ,IOT/M2M

The similarities with the mentioned technologies are that they employ small cell architecture which are deployed at the customer side or in the sensing environment.In the case of 5G,dense small cell networks are to be deployed.5G will require improved and new network paradimns.

There is a general agreement that in 5G,the nature of network functions will be fundermentally different than the previous evolutions that have taken place in mobile communication network functions in relation to 5G will relate not only to aspects of connectivity but to that of computation as well, but also computation and storage in the 5G segments. The network funtions provided include services such as filtering, forwarding and packet inspection. These functions will be at the edge or inside the network. Amongst the classes of network functions in 5G will be Virtual Network Functions (VNF). These are represent by one or several virtual machines running different software and processes on the various networking infrastructure.

In traditional networks, these functions are implemented by specialised or dedicated hardware (routers firewalls etc), however in this case they are implemented as virtualised functions. The other characteristic of network functions is the support of very diverse services which have varying and different requirements. These maybe enabled by certain network functions dedicated to these services or parametised to suite the different service requirements.

In general network functions are to be mapped to the physical architecture which highly depend on the use case,the service requirements and the physical properties of the existing networks. As a result a coexistence of different use cases in the same network implies the use of different VNF which are allocated within the network.

Two kinds of latency, User plane latency and Control plane latency. when a source node sends packets to the sink nodes,the packet experiences latency. This latency is experienced on the air interface and inside the node due to queing delay. Its is defined as as the one way time it takes to successfully deliver an application layer packet for a particular service in unloaded conditions. For 5G networks according to ITU-R, the minimum is 4ms for enhanced mobile broadband (eMBB) while its is 1 ms for Ultra-reliable and low-latency communications (URLLC), for user plane latency. The minimum requirements for control plane latency is 20ms whilst its is being encouraged by the proponents to lower to the control plane latency like 10ms. Control plane latency is defined as the transition time from one state for example Idle state to the start of continous data transfer i.e Active state.

1.5.2 Long Range (LoRa)

There are various IoT technoogies that are currently on the market. These include Sigfox, NBIOT and LORA amongst others. In this section,LoRa is discussed breifly. LORA is a good candidate for Virtual Buffer Extension Integration based on its simplicity and use of very small buffers. The principle idea is that the preamble time is split into two, one half for normal synchronisation and the other half for the time interval for the buffer contents to be sent to the sorrounding nodes when implementing virtual buffer extension. LoRa is a proprietory modulation scheme developed by Semetech,a US based company and it can achieve a sensitivity of over 148dBm with low cost of the bill of materials. This high sensistivity makes it very suitable for applications requiring range and robustness.

Some of the typical applications of LORA include Automated reading ,Home automation ,Security systems,Industrial monitoring and control as well as Long range irrigation systems.

LoRa stands for Long Range and uses Spread spectrum which is capable of achieving significanlty longer range than the existing systems which are based on Frequency shift Keying (FSK) or On Off Keying (OOK) modulation. Most of the devices have configurable Spread spectrum modulation Bandwidth (BW), Spreading factor (SF) and the error correction rate (CR) as shown in Table 1.2. These parameters can be set by the device depending on the deployment scenario. Each SF that is used in LoRa is orthorgonal, as a result multiple transmitted signals can occupy the same channel without interfereing. This allows the coexistence with other FSK based systems.

LoRa specifies a Phy layer technology, while the specification for the network and upper layers is maintained by the Lorawan alliance which is a group of different companies. The Lorawan specifies a standard interface to LoRa based communication systems. The LoRaWan standard specifies the message formmats and the timing for the uplink and downlink exchange of packets.

1.5.3 Spreading Factor

The SF in LoRa devices ranges from 6 to 12 as shown in Table 1.1, whilst the bandwidth show in Table 1.3, for example with the RFM97 device, bandwidth ranges from 7.8khz to 500Khz. In LORA, each bit of payload information is represented by multiple chips of information. The symbol rate,R_s is the rate at which information is spread ,the ratio between the symbol rate and and the chip rate is the SF which represents the number of symbols sent per bit of the information. Table 1.1 gives the range of spreading factors.

1.6 Energy Effieciency

Energy efficiency is the capability to minimise the radio access energy connsuption considering the traffic condition. While energy efficiency for a device is the

Table 1.1 Spreading Ranges.

Spreading Factor	Spreading Factor (Chips/Symbol)
6	64
7	128
8	256
9	512
10	1024
11	2048
12	4096

Table 1.2 Coding Rates.

Coding Rate	Cyclic Coding Rate (Chips/Symbol)	Overhead Ratio
1	4/5	1.25
2	4/6	1.5
3	4/7	1.75
4	4/8	2

Table 1.3 Bandwidth.

Bandwidth	Spreading Factor	Coding rate	Data Rate (kbps)
7.8	12	4/5	18
10.4	12	4/5	24
15.6	12	4/5	37
20.8	12	4/5	49
31.2	12	4/5	73
41.7	12	4/5	98
62.5	12	4/5	146
125	12	4/5	293
250	12	4/5	586
500	12	4/5	1172

capability to minimise the power consumed by the device in relation to the traffic statistics. The following two aspects are considered for energy efficiency for the network and the device:The efficient data transmission in a loaded case and the low energy consuption when there is no data.There are a number of emerging technologies and architectures in the IOT space and various device manufactures.I n the Low power Wide Area Network (LPWAN) space includes NB-IOT,LoRa and Sigfox . Also found are enablers of IOT such as Radio ferquency Identification (RFID) and Near Field Communication (NFC) technologies.The common tecnologies have been using cellular which includes GSM, GPRS, 2G ,3G and LTE as well as NB-IOT.

1.7 Energy/Power Consumption(State Based Energy Consumption)

This and subsequent sections discusses how modules are implemented and function in Omnet++ descrete event Simulator which was used for the simulation in this thesis.The energy consumption module is composed various parameters which describe the power consumption depending in which state the node is in. The states of the node are as follows:

(1) Reception state

(2) Transmission State

There are various signals that are required in order to trigger the energy module. These are Reception State Changed Signal under the reception state, the transmission state changed signal , under the transmission state. each signal is a composition of different parts each which have their own energy consumption requirements. The signal parts include Preamble part, Header part and Data part. These three compose or make up a signal. The other signal that comes to the module is related to the which signal part is being handled.We have the following:

(1) Received Signal part changed signal

(2) Transmitted Signal part changed signal

The other significant module is the mode of the radio unit. The radio is modeled to have three operating modes which are synonymous to the state of the radio. The modes are

(1) Receiver Mode

(2) Transmitter Mode

(3) and Transceiver Mode.

Under the the mode type, there is the radio mode changed signal. These are the various signals which trigger the radio mode in order for power measurements to be calculated. In order to calculate power, there are various parameters which are available as predetermined values.These can be described as follows: Sleep power consumption,switching power consumption,Idle power consumption,Busy power consumption,receiver receiving power consumption,receiving preamble power consumption,header power consumption and finally receiving data power consumption.Clearly if the Preamble,header and data are fixed, then the power consumption is deterministic. On the transmission side there is the transmitter idle power consumption,transmitter transmitting power consumption , transmitter transmitting preamblepower consumption,transmitter transmitting header power consumption and transmitter transmitting data power consumption. The energy module is a c++ class, when initialized has a pointer to the radio module in which it is located, which is the parent module.Once the radio module wherein lies the energy consumption module is established ,the radio module subscribes to various signals which act as a trigger. These signals are radio mode changed signal,reception state changed signal, transmission state changed signal,received signal part changed signal and transmitted signal part changed signal.

The energy module has a parameter which points to the type of energy source. In the "Inet" model in Omnet++, there are a number of energy sources which are available for use.These are categorised as energy storage. These are Idea energy source,Simple current battery Storage, and Simple energy based storage. The Ideal energy source store an infinite amount of energy and therefore can neither be completely discharged nor fully charged. Simple current battery Storage makes use of current computation and parameters in determining the power that is consumed and the simple energy source or storage maintains a residue energy capacity from the total energy that is consumed by the node.

When the energy capacity reaches zero,this node initiates the node crush or shutdown. When a node shuts down when power has been depleted, if it is part of a route,this means that the route ceases to exist.

1.8 Energy Source /Battery

In this thesis,the Simple energy module is adopted as the model for power consumption evaluation.The module consists of the following parameters:

1. Nominal capacity
2. initial capacity

The unit of measurement is Joules for the above parameters. It must be mentioned that this module does not model various other properties like self discharge, memory effect , over charging, temperature dependence etc that real world batteries would have.

The range of the nominal capacity is from 0 - infinity Joules, whilst the range of the residual capacity is from 0 - nominal capacity. The simple energy source employs the use of a timer which indicates the earliest time when the energy storage would be depleted. This timer facilitates the charging of the simple energy source , in this instance a battery. Once the timer expires, the variable target capacity is used to indicate how much the battery should be charged. Just like the energy consumer module, the simple energy module contains pointers.The pointers are the Node and the Node status pointers.

The Node pointer indicates the network node which contains the simple energy source and the node status pointer indicates the current status of the node. The simple energy source provides functions to be queried by an external module,these functions are:

1.Querying the Nominal capacity and;

2.Querying the residual energy/power.

In order to simulate the battery being connected, the module makes use of the function or method add energy consumer. For example in the real world a battery is connected to the terminals of a wireless router or inserted in the battery slot. This is represented by the method or function addEnergyConsumer. The other other functions are remove energy consumer, which represents the battery being disconnected,add energy generator ,representing charging of the battery, remove energy generator representing disconnecting of the charger. The charger can for example be a solar panel or charging from the mains. However in the case of offgrid nodes, this reperesents charging from solar panels.The other function is the receive signal function which represents a signal from a power consumer sent to the battery. These are the functions or methods which can be called by other modules in the node.

1.9 Consumed Power Algorithm

Within the node, there are various modules which can be added to enhance communication. For example, a wireless network node having only one network interface may have an additional slot to add another wireless interface. The existing wireless interface can be Wifi,for instance,and then the additional added slot can be Wimax. The new Wimax module has its own power requirements for it to function properly. These additional modules which consume power are classified under Energy consumers. Since various energy consumers can be connected to the energy source or battery, the c++ class implementation of the battery has a vector/list of type of the energy consumers and thus stores the list of energy consumers which are connected to the battery. In order to compute the total power, the battery class function loops through the list of connected energy consumer modules and obtains the power consumed by each energy consumer module to obtain the total total power consumed.

1.10 State Based energy Consumption

The energy consumer first receives a signal which can be any of the five mentioned (priously) above. On receiving a signal],the energy consumer module does the following:

(1) computes power consumption

(2) Emit a power consumption changed signal

The received signal is *radiomodechanged ,receptionstatechanged, transmissionStateChanged ,receivedpartchanged* or*transmittedpartchanged signal*. If the received signal is unknown,an error occurs.The energy consumer module first obtains the radio mode using its pointer to the radio it is associated with and does the following:

1.Determine current radio mode;

2.Based on the operating radio mode,which can be off,sleep or switching,return the corresponding power consumption.

If none of the above radio modes is obtained,the powerConsumption variable is initialized to zero.And this means the radio is receiving, transmitting or in tranceiver state. depending on the given mode, the module obtains the reception state or transmission state.The receiver can be in the following states RXIdle,RXbusy and returns the corresponding power consumption for that particular state.If in the receiving state,the received signal part is obtained, this can be the following: none,whole signal,preamble,header or data.

Similarly when the mode is transmitter or tranceiver,the transmission state is obtained.

1.11 Mac layer Implementations

The basis for the mac layer in Omnet++ Inet model is the mac layer base through which all implementations of the mac layer are subclassed.The basic maclayer base performs the following functions.

(1) Create an interface entry

(2) Add created interface entry above into an interface table

(3) Get the module in which this module is in and then register the interface.

The mac layer has four gates of inputs into its module, these are Upperlayer In, Upperlayer out, Lowerlayer in and lowerlayer out. Each gate in Omnet++ is assigned a gate id.In order to check the type of message that has been received, the messages have a get arrival gate id function whose result is compared to the id of one of the gates to determine weather its an upperlayer message or a lower layer message.

When a mac layer protocol is created or designed,a matching interface would have to be created as well.The mac interface subclasses from the wireless interface. It has the following parameters, radio type name and mac type name. These are given as parameters radio.typename and mac.typename.

The maclayer base also has a mac header base message from which implemented mac header messages are subclassed.A typical mac header contains source address, destination address and the network protocol id.

1.12 Finite Field Arithmetics

This section provides a brief review on finite fields as they are used in the design of our network coding module in this work. To understand finite fields, we take a look at groups. Finite fields are also known as galois field named after the french mathematician Evariste galois. Finite fields are used in many areas of computation in computers due to their mathematical propeties. One of them is that they solved the problem of register overflow. Since registers in computers use 8 bit,computation which resulted in more bits resulted in register overflows. A finite field is defined as a set which has a finite number of elements.The operations that can be performed in a finite field are addition, multiplication, subtraction and divition(inversion). In order to understand fields,an understanding of groups is required , which a brief introduction is given below.

A group can be defined as a set $\mathcal{G}$,which has an operation * combining two elements of its set.Groups have the following properties or axioms:

(1) The group operation for all $x, y \in \mathcal{G}$, $x * y = z \in \mathcal{G}$

(2) The group operation is associative

(3) The group contains an identity element

(4) The group contains an inverse element for each element of the group.

(5) The group is abelian or commutative ,if x*y= y*x for all x,y in G.

A group contains only one operation on its set and the corresponding inverse operation. If the operation is addition ,the corresponding inverse operation is subtraction.And if the operation is Multiplication,the corresponding inverse operation is division.

A field on the otherhand has all the four basic arithmetic operations namely addition,subtraction,multiplication and division.The following are the axioms or properties of a field:

(1) All the elements form an additive group with ath addition operation,"+".

(2) All elements ,excluding 0 form a multiplicative group,and the operation is "X",and the 1 is the neutral element.

1.13 Assumptions

The first assuption is that during evaluation,the node positions remain fixed as packets traverse the network end to end from the source nodes to the Sink nodes. Limited size of the buffer is assumed and also limited battery size is also assumed. Once the battery in a node is depleted,the node is out of commission and all previous routes that pass through that nodes are all lost and the route is unavailable.

1.13.1 Related Literature

The authors in [17][18] study distributed storage architectures and a distributed storage model and a layered access control model which is distributed is developed. In [**?**] repair for the damaged storage node in the distributed storage system is considered. Work in [20] simulates distributed storage systems using network coding and the studies in [21] focus on how to better cope with node dynamics and failures. Authors in [18] propose a new infrastructure of storage overlay networks (SONs). This work is related to [23][16] in which buffer-aided physical-layer network coding (PLNC) techniques for improving data transmission. A network coding approach is used for mobile cloud storage, however in our work we look at high priority packets and latency and packet reduction.

1.14 Thesis Structure

The structure of this thesis is as follows and illustrated in **??**. This thesis consists of 7 chapters . Chapter One provides the Introduction to this thesis , stating the motivation and aim, this part exaplains the driving force for the subject matter. Also related technologies are reviewed. Chapter 2 provides background information on Network Coding and how it relates to the network topology. The algebraic approach to network coding is examined,review on network coding concepts using an algebraic approach and takes a look at the multicast problem in Information transfer . Chapter 3 covers routing considerations with multiple source -destination routes . Chapter 4 covers Implementation of a network coding module in simulation. In chapter 5 investigates the feasibility of a protocol for network coding opportunity discover for relay nodes while Chapter 6 covers buffer extension in Multi-hop networks for packet delay improvement.And finally chapter 7 is the conclusion and future works.

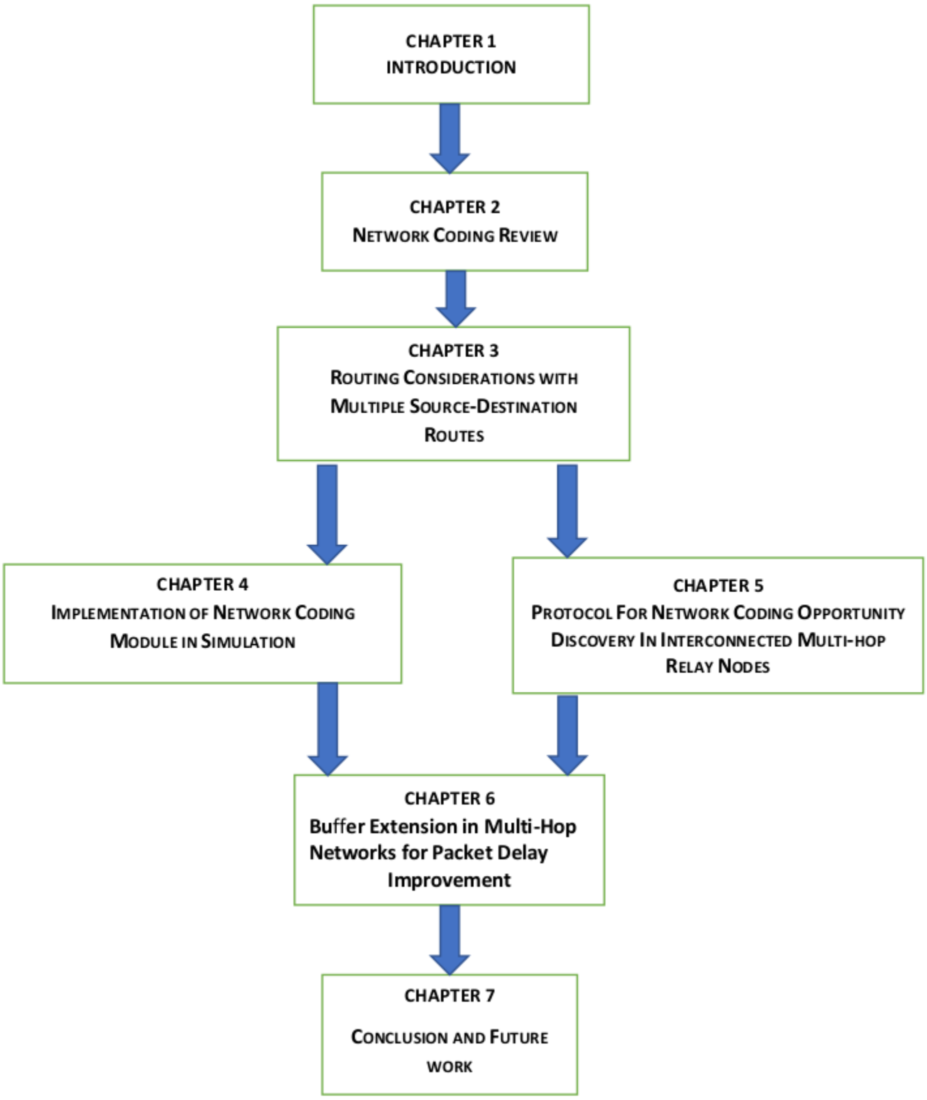

Figure 1.1 Flow of this thesis

Chapter 2

Network Coding Review

2.1 Classes of Network Coding

Network coding can be divided into two broad based categories ,Physical Layer Network Coding Network coding at the Network Layer.Furthermore,it can then be subdivided into Linear Network Coding and Random Network coding which is explained in the below.Network coding is described in the seminal paper [1].It describes the now popular butterfly network in which packets are multicast from a group of source nodes to a group of Sink nodes.Phy layer network coding involves adding the waveform at the physical layer hence improving on throughput.

The theory behind network coding is thatwhen a packet network is considered and instead of simply routing packets,intermediate nodes can compute and forward functions resulting from such a computation of the received packets which enter the node's input and exit using the output links.Traditional packet networks at the intermediate nodes implement a Store and forward technique of packets.Meanwhile when network coding is considered, the nodes Compute and forward the packets.

When a butterfly network is considered,a single source transmits information to two destinations or sink nodes which is called multicasting.In network coding ,the channels or interconnecting links in the network which link various nodes are considered not to experience anty error in packet transmission from one node to another .The network experiences a bottle neck on the interconnecting links.When considerations is made to routing and the issue of sending data or packets to the two sink nodes,routing succesfully delivers three packets in two time slots thereby achieves a multicast throughput of 1.5 packets per channel use.

On the other hand ,network coding solution on the other hand achieves an improved multicast of 2 packets per channel.This is achieved by perfoming modulo two addition of packets at the bottle neck link.As a result the throughput improves at the expense of an encoding operation either at the source or intermediate nodes as well as decoding operations at the destination nodes.The network coding solution provides the optimum multicast rate that can be achieved.

The key lesson obtained is that routing alone in general is insufficient to fully utilise the information carrying capability of communication networks.

2.2 Multicast Problem

This section is a review based on insights obtained for the works in [26].Given a network denoted by the graph $G = (V, E)$ consisting of a set of vertices V and a set of edges E,and has a sink node denoted as $t \subseteq V$ which can be reached via the graph G from another vertex $s \subseteq V$.

A set of the edges formed is made up of aroute which runs from s to t,if these links are broken,the resulting effect is the isolation of s and t. The smallest number amongst all the possible cuts from s to t which results in complete isolation of the two vertices is termed the minimum cut. This cardinality of the s,t seperation is the smallest possible number of links that can disconnect s and t completely.This point provides a bottle neck to the packets flowing from s to t.This bottle neck limits the flow of information between s and t.The information flow from the source to the bottle neck and from the bottle neck to the sink.From the data-processing inequality which is derived from classical information theory we have the equation 2–1 depicted as:

$$I(R(s,c)) \leqslant R(c,t) \tag{2–1}$$

where $I(R(s,c))$ is the mutual information between the source s, and the sink t which cannot exceed that between the source s and the bottle neck point c,which is the cut point,thus

$$I(X_{O(s)}; X_{I(t)}) \leqslant |C| \tag{2–2}$$

where C represents the the s-t seperating cut.

$$I(X_{O(s)}; X_{I(t)}) \leqslant mincut(s,t) \tag{2–3}$$

$X_{O(s)}$ in 2–2 and 2–3 are the packets at the output of source s and $X_{I(t)}$ are the packets at the input of the sink node t.The above is the upper bound obtained when $|C|$ is minimised.

Therefore the information rate between s and t is depicted in equation 2–4:

$$R(s,t) \leqslant mincut(s,t) \tag{2–4}$$

From the theory of commodity flow,it is well know that the minimum number of pairwise edge-disjoints routes from *s* to *t* is equal to the maximum number of pair wise edge-disjoint routes from *s* to *t*.These edge disjoint routes can be found using well known algorithmssuch as the Ford-Fulkerson algorithm.Given that the number of destination nodes has more than one sink nodes,the multicast rate to all the nodes cannot exceed minimum of the transmission rate from the source to any of the nodes in the destination.

Given a set of sink nodes T,having multiple nodes in it,the multicast rate from s to T must satisfy the equation 2–5 which is obtained as:

$$R(s,T) \leqslant min_{t \subset T} mincut(s,t) \tag{2–5}$$

which is achievable via network coding.

2.3 State Space Approach to the Packet Network

In a state space description of the network,the state of the network is described by the packets that are being transmitted by the links in the network.This can be given by equations 2–6 and 2–7 which are depicted as:

$$X = AX + BU \tag{2–6}$$

$$X_{I(t)} = C_t X \tag{2–7}$$

Where A is the link to link transfer matrix or the adjacency matrix,U is the source packets,B is the matrix representing the connection of the source packets to the network in equation 2–6 .Matrix A is common to all the receivers and shows how the links interconnect to form a particular topology as the one illustrated in Fig. 2.1.

An element of A is non zero if there is node connecting the two links.Thus the element can be either equal to 1 or to an indeterminate in ai,j.Network code

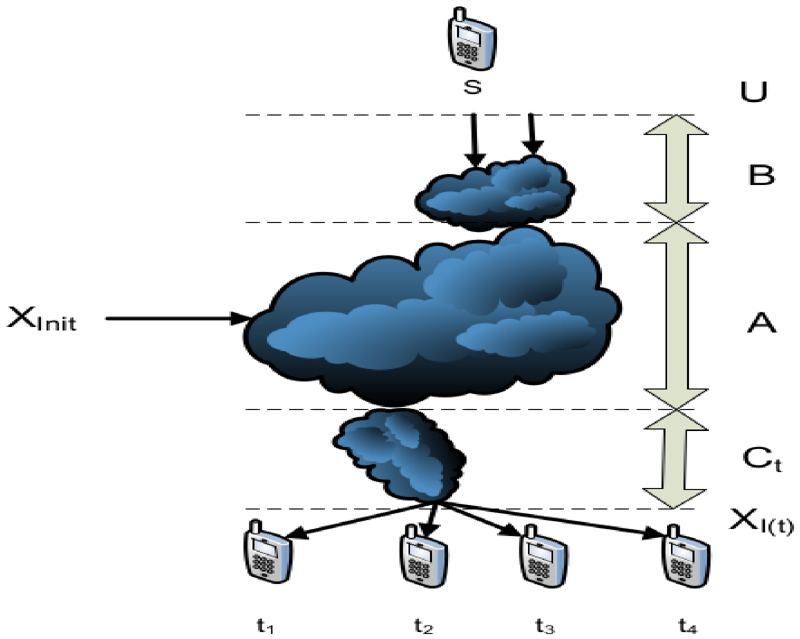

Figure 2.1 visualisation of network coding theory state equations

design thus amounts to selecting the indeterminate entrie in matrix A.The matrix A is nil potent for an acyclic graph.Which means there is a positive integer L such that $A^L + 1 = 0$.This translates to equation 2–8:

$$(I - A)^{-1} = I + A + A^2 + ...A^L \tag{2–8}$$

which forms a polynomial in A.

$$G = C(I - A)^{-1}B \tag{2–9}$$

The equation **??** shows the global tranfer matrix that is obtained or seen at each sink node.In order for the network coding scheme to function and provide a multicast rate to all the nodes which make up the sink nodes,the matrix G should have not have a zero determinant.Therefore the product of the determinants of the global matrices seen by the sink nodes should not be equal to zero for succesful decoding of packets.Fig. 2.2 generalises the network coding process as previously discussed.

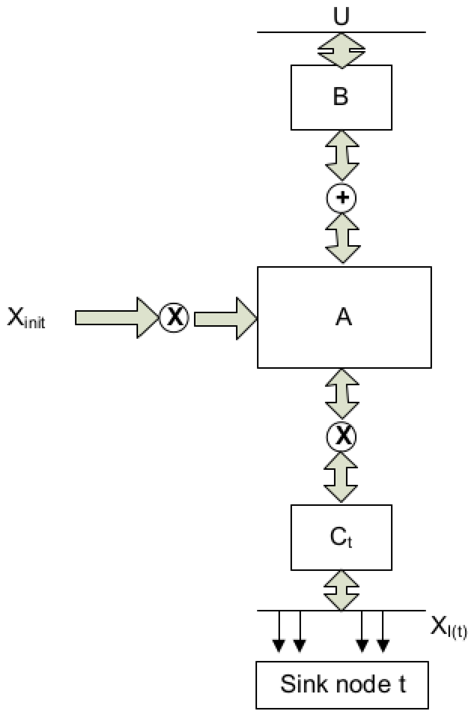

Figure 2.2 Block diagram of Network Coding

2.4 Network Coding and Routing

In the case of hybrid schemes,instead of the intermediate nodes performing the coding of packets,coding is done at the source node and the encoded information is sent out into the network.In this scenario,the intermediate nodes act as relay nodes and forward the packets to the nexthop nodes in the network.In the case where the network is assumed to have no losses,the maximum rate can be maximised by combining with routing.Packets are coded at the source using network coding and then routed using existing routing protocols,this depend on the type of application.

Chapter 3

Routing Considerations with Multiple Source-Destination Routes

Routing is a cardinal part of current networks which enables communication from one node to the other in a network.Through the use of routing protocols,packets are routed from a source node to a destination node.Routing is a functionality of the network layer. IPv4 or IPV6 herein after called IP performs the functions of creating packets in the network layer.A network node contains an interface table where all interfaces which are available within the node are registered.This ranges from ethernet interfaces ,wireless interfaces and various other interfaces which are used in communicating with the outside world. Apart from the interface table, a node contains a routing table where route entries of ip addresses to various destinations are stored.

When the network layer receives packets from the upper layers,it encapsulates the datagrams coming from the higher layers and then sends them to the lower layers for further processing,at this stage ,the IP looks up the network address specified in the datagram.In most networks this address is specified by the application layer which is the source of the data and addresses are appended in the datagram.The IP,using the destination address specified in the packet from the upper layer queries the routing table for a route.The routing table holds the routes which are generated by the routing protocol being used by the network node.There are many routing protocols available such as Open Short Path First (OSPF),Dynamic MANET On-demand(DYMO),Ad hoc On-Demand Distance Vector (AODV) routing protocols etc which work in conjuction with the network layer for the purpose of routing packets.

Based on the address in the datagram from the upper layers,the IP queries the routing table by for the route to the destionation.If the entry is present , then the route is present. Most current routing protocols only maintain a unicast path in the routing table.

3.0.1 Changes to enable storage of Multipaths in a Routing Table

Most routing protocols compile a list of possible routes during routing discovery and select the best among them based on a criteria such as the shortest path,and discards the rest.Instead of discarding the alternative routes ,they are instead kept and ranked based on additionnal metrics such as available battery power in the route to support the given session of communication.The paths can then be weigthed and then ranked using additional metrics can such buffer size and battery power.Due to the dynamic nature of the buffers and battery power,its is difficault to know for certain at any given time the exact battery values especially in a highly dynamic environment.For example,during route discovery the battery level information can be attached to the the RREQ for battery information to be provided to the destination node for the return path.And the RREP for the return path can carry battery information to the source node.The use of control information is as illustrated in Fig. 3.1.The figure illustrates the logical path in a multihop network ,not necessary a physical path as illustrated in the figure.Node TX is the source of the packets,node 1 and node 2 are the relay node and Node RX is the destination node.When node TX has a packet to send and it does not know the route,it is assumed the diagram depicts the logical route in a mesh network which a packet would take from source to destination.At time T1,the node TX sends a RREQ,as the RREQ traces multiple routes,it collects battery information in its header which can be used at the destination end.The power level,which can be inform of the energy residue of the battery is recorded as the control packet is relayed.The relay nodes in this case are node 1,node 2 upto node N where N is the number of intermediate relay nodes between the source and destination nodes route.At time T1,node TX sends an RREQ packet to request for a route to node Rx,During this time,the relay nodes record the energy value of the battery.At time T2,once the destination node is reachable,Node RX replies whith an RREP,and on the route back to the source records the value of the energy at each hop.At time T3,once RREP packet is received by nodeTX,it sends the data in form of packets to the destination.

The challenge that comes with using the energy values in such a fashion arises in the case where the relay nodes receive packets from more than one source

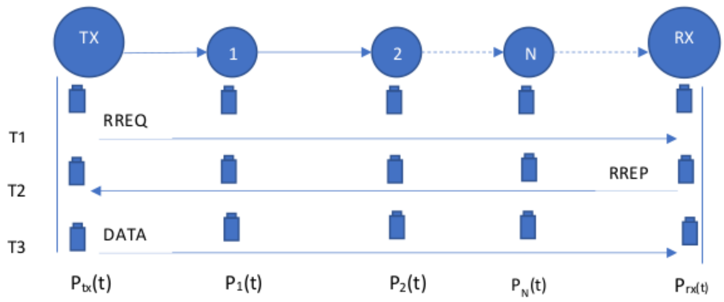

Figure 3.1 Collecting battery info using routing control packets

i.e if the node acts both as a relay node and as an access node,by the time the control packet RREQ reaches the destination node,the relay nodes would have relayed new packets from its clients or from other source nodes,hence by the time the RREP packet pases the same relay node from the node RX,the power values would have changed.If The nodes do not have rechargeable power,then once a node has depleted power and shuts down,the routes no longer exists hence an alternative route needs to be used.

3.0.2 Multipath Routes

There are several ways in which multipath routing is used to achieve dependability.This includes:

1. Alternate paths are maintained,when one path fails,the other one takes over.In this case one route is used and the other routes are on standby.This is commonly denoted as N+1 redundancy,where N is the number of active or used routes and 1 being the redundant.

2. By using a scheduling strategy among the multipaths available,data allocation is chosen which maximises the probability of succesfull reception.By using a proper scheduling criteria,the chances of transimitting data succesfully increases as well. Given the duration for a packet to be transmitted is much smaller than the interval of the topology change,then it is assumed that the topology will not significantly change while the packet is being transmitted.In this thesis,it is assumed that the paths are mutually disjoint meaning the routes share no common nodes from source to destination.The Multiroutes from node S to node D as shown in Fig. 3.2 are Route : 1 S-1-2-3-D, Route 2 : S-4-5-D and Route 3 : S-6-7-8-D.

Given the different multipaths in the network from source to destination,the

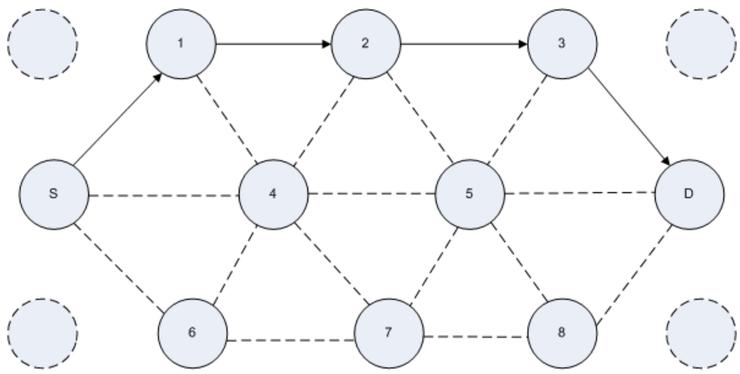

Figure 3.2 Multiple Routes

various paths are ranked based on the probability of successful transmission p_i .These probabilities are ordered from the best available path to the least performing paths.This is as shown in 3–1 generalised as:

$$p_i \leqslant p_{i+1} \tag{3–1}$$

A path ranking algorithm can then be used based on the probability of succesful transmission from source to destination for all the paths or routes.The probability of failure for each route considering Buffer size and Battery Power can be generalised as in equation 3–2 as :

$$P_f ail = 1 - p_i \tag{3–2}$$

where P_{fail} is the probability of failure which include the probability of the buffer in a relay node being full or the battery power being depleted.

3.0.3 Route Selection

Different intermediate nodes in the network have different buffer sizes.The data rate in each route from source node to destination node will be affected by the bottle neck node the whole route adopts the minimum data rate between two nodes along the whole route. Given the network topology is a multi-hop mesh configuration as illustrated in 3.2,the burden to apply intelligence is on the sending node with assistance from the receiving node in the network.When a multicast scenario is considered with a single source and several sink nodes,applying a network code ensures that the sink nodes all receive the sent data at the same rate ,r.Given that there are r data processes in the source node,it translates to there being r node disjoint paths or routes from source to each of the sink nodes .This is based on the assumption that the capacity of each arc or link is 1 symbol per unit time.

In the first instance,the intermediate nodes do not participate in the route selection.Then in the second instance,the intermediate nodes can participate in route selection. What then is the criteria that is used for route selection in a multi route scenario?In conventional systems,routes are stored in routing tables and chosen based on a certain routing criteria.Algorithms such as Djikstra calculate the shortest path to a target node whilst the ford-fulkerson algorithm finds the number of disjoint paths from a source node to a destination node.

In conventional methods,the routing is static or dynamic.In mobile adhoc networks,there is a rapid change in the network topology resulting in routes being valid for only a limited amount of time.Once the topology changes ,the destination nodes may become unavailable when there is only a single route.The case is different for multiple routes from a source to a given destination.

Given that there are r routes from source to destination ,when one route breaks,then the amount of routes remaining becomes

$$numberofroutes = (r - 1) \tag{3–3}$$

consequently,the data rate drops to (r-1) symbols per unit time.This drop in data rate means that the min-max flow is affected as well and it becomes:

$$min - max = (r - 1) \tag{3–4}$$

It therefore follows that the rate to (r-1) symbols per unit time.There are several cases that will cause a route to cease to be available.Topology changes,Battery issue,packet loss.Topology causes route changes due to the mobility of the nodes.when a node which is part of a route to a destination moves,it breaks the connection that it has with the surrounding nodes causing that route to be invalid.This therefore means that some routing algorithms are constantly updating the state of routes,however if there are rapid changes in the topology,then the updates increases which can affect the energy of the nodes.

3.0.4 Buffer and Power Considerations

When power is depletion is considered in a route,it only takes one node in a path to the destination node for that route to be invalidated .Packet losses due to buffer being full may be only temporary depending on how long the affected node will be receiving heavy traffic.There a number of strategies available to prevent buffer overflow by use of mac protocol techniques.In this thesis it is assumed that the topology doesn't change during the time the packet is being transmitted from source to destination.With all the factors which can affect the route mentioned above,the sender node has to calculate the most suitable routes based on the network condition.

3.0.5 Node Disjoint Routes

Let the number of node disjoint paths paths from source to destination be given by n.In essence this is equivalent to having n number of source processes at unit rate ie source node has processes $X_1, X_2...X_n$. When a route becomes invalid,in order to maintain the Network coding theory main algorithm for multicast networks the number of source processes need to match the available edge disjoint routes to the destination or an additional route to the destination has to be found in order to match the source process to fufil the network coding criteria.This is a scenario in which the number of source processes is greater than the number of disjoint paths to the sink nodes.

3.0.6 Static and Mobile Networks

A static network with any of the above factors causing route unavailability maybe equivalent to the problem that is faced by routing in mobile networks with nodes which frequently change their positions.For example in a fixed network if a battery is completely depleted,the node shuts down and becomes inactive,this changes the topology of the network in terms of connectivity. The underlying factor is that they all affect the topology and thus affect the available usable routes.In traditional telecommunication networks ,several methods are used for trunk reliability.The strategies include trunk reservation,virtual channel protection and the use of a transiting exchange to reroute the affected traffic.During link overload conditions, a reservation parameter allows access to the transiting exchange trunk if there are idle links available. The downside is that it is a local strategy and considers only 1 trunk group,and not the entire end to end connection.This one way mechanism protects traffic from other nodes but not the other way round.Mutual protection of connection and services is required depending on the QoS strategy that is implemented.It is Assumed the sending node is aware of the route states ie when congestion occurs the sender is notified to make a routing decision.

Chapter 4

Implementation of Network Coding Module in Simulation

The network coding module implementation is located in the network layer of the OSI/IP layer .This is as shown in Fig. **??**.The figure shows the various layers of the TCP/IP layer which include the link layer,network,transport and Application layer.The simulation used is Omnet++ and in order to implement the network coding module,the inet model for network nodes is used.In the model it communicates directly with the IPV4 modules of the network layer.Packets from from the network enter the node through the Wireless network interface.In order to model Multiple paths from source to destination,several wireless network interfaces can be used as shown in the figure.These are labelled wlan[0],wlan[1] and wlan[2].

In the case of implementation using the wireless network interfaces as LoRa modules,the node can then communicate with a gateway on a particular spreading factor (SF).Each wlan can have use a different spreading factor.The spreading factors in LoRa are orthogonal to each other hence they do not interfere with each others.The wireless network interface abstracts the Physical layer or Phy layer implementation.The from the wlan are sent to the network layer through the to the IPV4 module.The IPV4 modules implements network layer functionality using Internet protocol version 4.

4.1 Finite Field Implementation

To perform network coding of packets in the network coding module,a finite field is implemented as a table from which the coding coefficients are obtained.The Finite field used is as shown in Table 4.1.The Galois field being used is GF(256).The table shows 256 entries which are the 256 elements generated us-

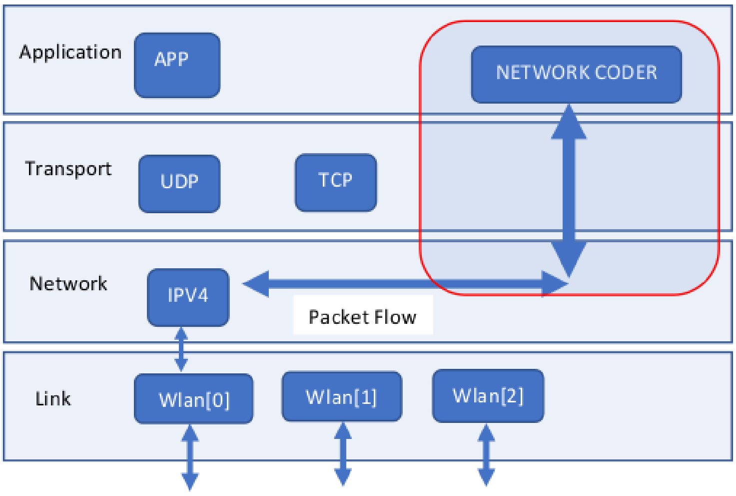

Figure 4.1 Network coding Module placement in a Network node.

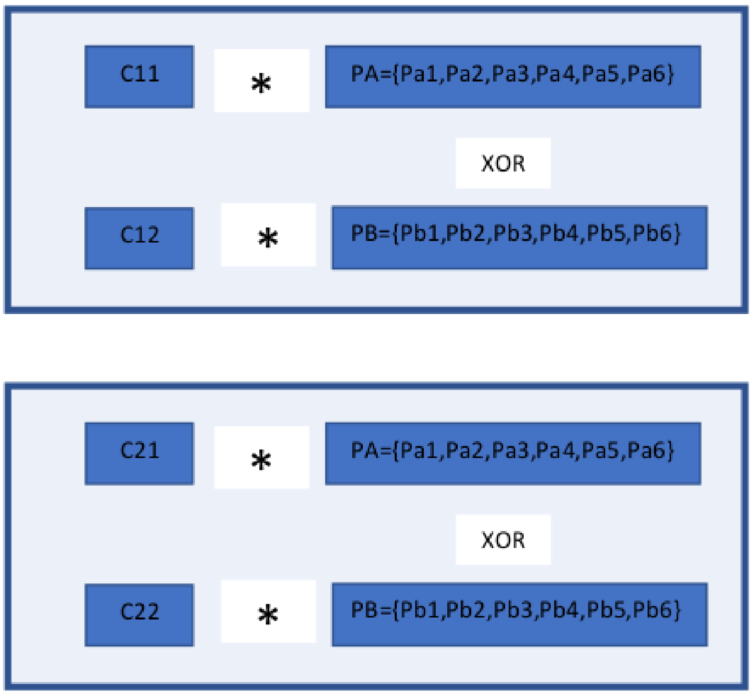

Figure 4.2 Coding in packets with coefficients from finite field

Table 4.1 Finite Field GF(256).

0-15	0	1	2	4	8	16	32	64	128	113	226	181	27	54	108	216
16-31	193	243	151	95	190	13	26	52	104	208	209	211	215	223	207	239
32-47	175	47	94	188	9	18	36	72	144	81	162	53	106	212	217	195
48-63	247	159	79	158	77	154	69	138	101	202	229	187	7	14	28	56
64-79	112	224	177	19	38	76	152	65	130	117	234	165	59	118	236	169
80-95	35	70	140	104	210	213	219	199	255	143	111	222	205	235	167	63
96-111	126	252	137	99	198	253	139	103	206	237	171	39	78	156	73	146
112-127	85	170	37	74	148	89	178	21	42	84	168	33	66	132	121	242
128-143	149	91	182	29	58	116	232	161	51	102	204	233	163	55	110	220
144-159	201	227	183	31	62	124	248	129	115	230	189	11	22	44	88	176
160-175	17	34	68	136	97	194	245	155	71	142	109	218	197	251	135	127
176-191	254	141	107	214	221	203	231	191	15	30	60	120	240	145	83	166
192-207	61	122	244	153	67	134	125	250	133	123	246	157	75	150	93	186
208-223	5	10	20	40	80	160	49	98	196	249	131	119	238	173	43	86
224-239	172	41	82	164	57	114	228	185	3	6	12	24	48	96	192	241
240-255	147	87	174	45	90	180	25	50	100	200	225	179	23	46	92	184

ing 4–1 as the irreducible polynomial.When expresses in decimal,the irreducible polynomial is 369.The leftmost column indicates the index of the element in the table from left to right along each row.For example the element with index 15 in the table is 216.The irriducible polynomial used to generate the galois field is :

$$f(X) = X^8 + X^6 + X^5 + X^4 + 1 \tag{4–1}$$

where $f(X)$ is the irreducible polynomial.

4.2 Network Coding

The packets into the network coding being used have a length of 255 bytes.The original packets are generated by the application layer in the APP module having a packet length of 223 bytes.An optional reed solomon encoder can be used.In this thesis the reed solomon encoder RS (n,k) used is the RS(255,223).

When two packets are coded at the network layer as shown in 4.2,the two packets are coded using coefficients C11,C12,C21 and C22 which are elements of a finite field.To perform the network coding,the module uses a two registers bank.Given two packets PA and PB.They are first put into the register and the contents are of each register are multiplied using coefficients which are uniformly drawn from the GF (256) table.Since these coefficients are randomly drawn from the the GF (256),this implementation is a random linear network coding.After multiplying,the packets are added using exclusive OR

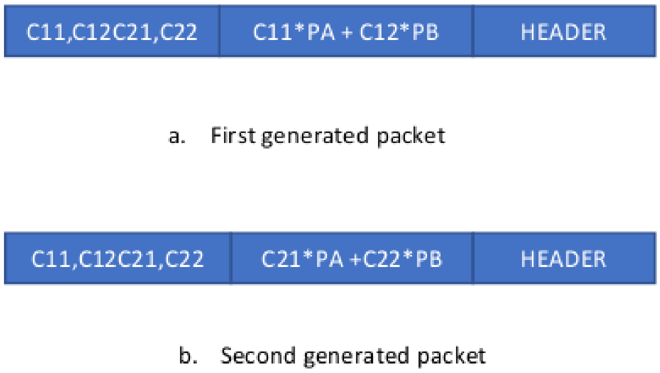

Figure 4.3 Coded Packets.(a) shows the first coded packet using a set of coefficients C11 and C12 and (b) shows the second set of coefficients C21 and C22.

(XOR) which acts as the coding function.

4.3 Network Coded Packet

The network coded packet which is sent to the sorrounding nodes or originates form the source node has a structure as shown in Fig. 4.3 (a) and (b).The coefficients used in the network coding of packets are attached to the packet before it is sent.In this figure four coefficients are attached which adds an overhead of 4 bytes.The figure also shows the result of the coded packet and the header which is appended at the lower layers.The appended coefficients makes it easy during the decoding process for the receiver or the decoding module to just extract the coefficients for decoding.

4.4 Conclusion of Chapter

This chapter shows how network coding is implemented in the simulation at the node level.A network coding module is implemented at the network layer which performs network coding of packets.The coding coefficients are obtained from a generated table of finite field elements in GF(256).Packeks leaving the node are appended with the coding coefficients and a header from the lower layers.

Chapter 5

Protocol For Network Coding Opportunity Discovery In Inter-connected Multi-hop Relay Nodes

5.1 Introduction

One of the challenges faced in practical network coding application is discovering of network coding opportunities for intermediate nodes to perform network Coding. This is because the number of coding opportunity has an impact on the performance of the network such as delay. On one hand, coding opportunities need to be discovered, on the other hand, the number of coding nodes in a multihop relay network have to be few to minimise delay. From the seminal paper in [1], the concept of network coding was realized. Adhoc On Demand Distance vector (AODV) routing protocol is described extensively in [2]. The Authors in [3] study converting an on demand protocol into proactive multihop routing protocol for link failure prediction. In [4], the authors propose a new enhanced AODV routing protocol, which is based on the most stable route between the source and destination node . To consider security concerns,[5] looks at a trusted AODV routing protocol whilst in [13], reliability of AODV is considered. This chapter investigates coding node discovery for its potential application in for inter-connected multiple hop networks and in interconnected wireless BAN to multihop relay networks through a mesh gateway.

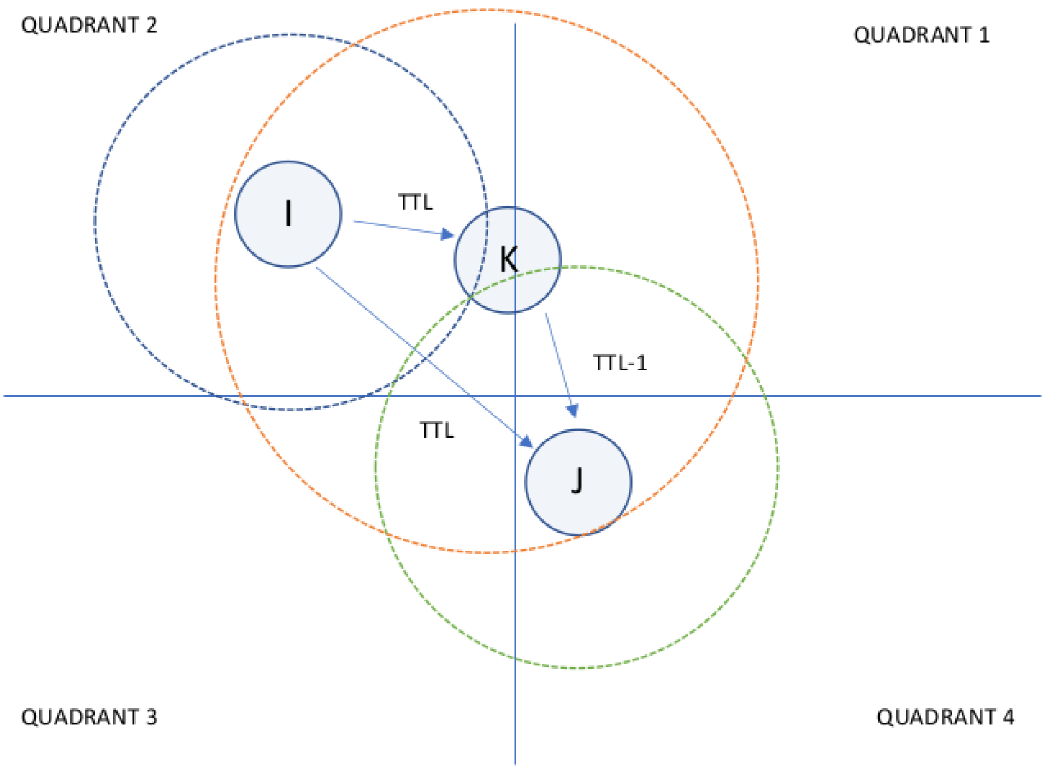

Figure 5.1 potential coding pair discovery

5.2 Current Network Node Discovery for Network Coding

The current method used in detection of coding opportunities works as follows: Node I transmits a packet and it is received by both node K and node J if they are within transmission range. Reference is being made to Fig. 5.1. When node J receives a packet from node I, it retransmits the packets with a unit reduction in the Time to Live (TTL), $TTL = N - 1$.

After node J rebroadcasts the packet from node I, node K receives this packet. The role of node K in the network is to act as a coding point, therefore it has to identify the coding opportunities in the network based on the received packets that it has to forward or relay. When node K receives the rebroadcasted packet, it compares the TTL of the original packet which it had received also from node I earlier. If the latest rebroadcast packet has a lower TTL than the original, then node K knows that node I and J are potential network coding pair

```
:BUFFERED COUNT : 4
:TTL VALUE : 33
:RECEIVED :  Xcoord : 264.041     Ycoord : 780.23
:GW COORD :  X :421.039           Y : 875.38
:ANGLE TO NODE (TAN THETA) : 0.60606
:QUADRANT : 2
:ANGLE :            148.7820         DEGREES
:INSERTING INTO MAP
:INSERT DONE

:IP                              QUADRANT
:----------------------------------------------------

:145.236.0.1                        1
:145.236.0.2                        1
:145.236.0.4                        1
:145.236.0.15                       2

:IP                              ANGLE
:----------------------------------------------------
:145.236.0.1                     82.87
:145.236.0.2                     22.5509
:145.236.0.4                     18.0112
:145.236.0.15                    148.782
```

Figure 5.2 Quadrants of IP

neighbours.This means that packets for Node I and node J can potentially be coded together for onward transmission to the rest of the network.

5.3 Protocol for Discovering Potential Coding Nodes

Node I ' s transmission can be heard by coding node K, but not node J. Node K can hear transmissions from both node I and node J. Since Node K can hear both transmissions, Node K suggests a pairing of the nodes so as to achieve network coding.Node K hears many transmissions from other nodes as well and therefore has to decide which packets to combine.

5.3.1 Packet Structures

If the direction of arrival of the packets is within a pairing threshold angle θ, then the coding node can suggest the pairing of nodes withing a small distance. This suggestion is in form of a control packet sent either to Node I or Node K from node J for example as shown in Fig. 5.1 based on criteria which can be determined such as node capability being battery power and or buffer size or processing capability. The control packet instructs the movement of the non neighbour node in order to be in the communication range of the other node resulting in forming a neighboring pair.The control packet to effect movement is structured as Movement Request (MREQ).The packet structure is shown in Fig. 5.3 and the detailed packet structure of the MREQ is as shown in Fig. 5.4.The MREQ packet is 24 Bytes and is based on the RREQ packet.Consists of Source address,Destination address ,Mreqid,Hopcount ,command ,urgency Xcoord,Ycoord and Reserved.Source and destination addresses are the Network Internet protocol addresses of the sending and receiving node respectively. Mreqid is the id of the MREQ packet,Hopcount is the count of the node the packet has traversed.Command encodes the movement details ,urgency how fast the node to move to the new position.Xcoord and Ycoord are the X and Y coordinates where the node should move to in order to pair with a neighbouring node to achieve network coding.The reserved field for other information which can be carried by the packet. The node receiving the command to change location has to perform its own cost benefit analysis.The steps are as follows:

- Node K obtains/Calculates location of node I and node J.

- Node K evaluates if node I and J are potential coding Neighbours. K obtains/Calculates location of node I and node J.
- Node K obtains transmission range of node I and node K obtains/Calculates location of node I and node J
- Based on a decision criteria such as mobility of the nodes, node K sends a request for node I or J to move. This is using packet as dipicted in Fig. 5.3 and Fig. 5.4 .
- Node K determines how urgent the node has to move based on delay constraint.
- Based on the urgency, node I or J determines the speed and moves into the other nodes range.
- Node K confirms that node I and node J are now a coding pair and can now perform network coding on packets from node I and node J going to a particular destination.

In order to obtain location information of neighbouring nodes, packets have an option to carry position information. The Node could also perform ranging,given it has ranging capabilities like those of Ultra Wide Band (UWB) to obtain the distance to the neighbouring node. Using AODV, location information of nodes can be exchanged during route discovery.This could be enabled by extending the route request packet with an additional 2 bytes field to carry the extra location information ,this is illustrated using Fig. 5.5.The total packet size of the RREQ becomes 26 bytes from 24 bytes.

Fig. 5.6 shows WBAN gateways interconnected to a mesh gateway which connects to a multihop relay mesh network.Node 2 acts as a gateway node which can interconnect with various other WBAN gateways 1,3 and 4 and provides access to the multi-hop or mesh network for transmission of medical data.There are four quadrants numbered one to four as well as border lines between the quadrants which are four which is as shown in Fig. 5.1. The gateway node can as a result keep the neighbour node id mapped to its quadrant and angle.Neigbours falling in the same quadrant can as a result be grouped together.

The mapping of node id into various quadrants is as shown in fig. 5.2.This is as seen from the perspective of the mesh gateway node.The figure shows that the number of buffered packets is 4 and the time to live ,TTL value of the received

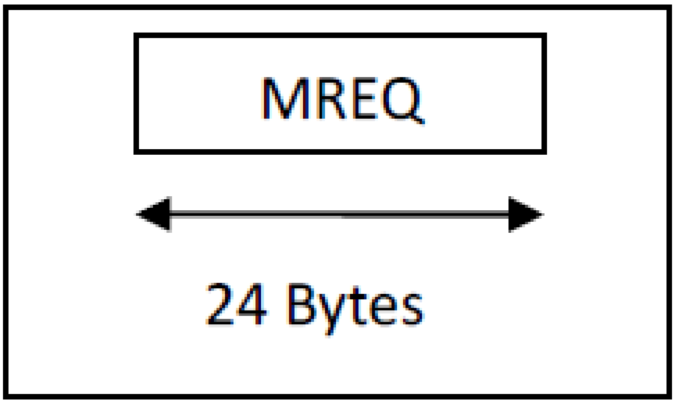

Figure 5.3 MREQ packet structure

packet is 33.Location information is extracted form the packet from the Xcoord and Ycoord parameters.Based on its own location,the mesh gateway calculated the angle or quadrant where the packet source is located.Various IP addresses are shown and mapped into quadrants 1 and 2 as shown and the angles as well.

The IP addresses in the same quadrant can thus be classified as neighbouring nodes and packets from them or addressed to them can be network coded.From the figure addresses 145.236.0.1 , 145.236.0.2 , 145.236.0.4 are grouped in quadrant 1 with angles 82.87 , 22.5509 , 18.0112 degrees respectively.Because the gateway node is connected to many WBAN nodes, the traffic intensity can be high.

The impact of RREQ and data from several WBAN gateway to the mesh gateway node is investigated in this section.Given that there are $N+1$ WBAN gateways attached to the mesh gateway, λ is the total traffic intensity from each WBAN gateway.Thus each WBAN gateway contributes λ_i.And the total traffic intensity is the summation of the contributions by the different gateways.This is shown using equation (5–1).

$$\lambda = \sum_{i=0}^{N} \lambda_i \tag{5–1}$$

$$P_B = 1 - \frac{(1-\rho)\rho^n}{1-\rho^{B+1}} \tag{5–2}$$

Traffic Intensity T1,is given by:

$$T_1 = \frac{\sum_{i=1}^{N} \gamma_i}{\sum_{j=1}^{N} \xi_i} \tag{5–3}$$

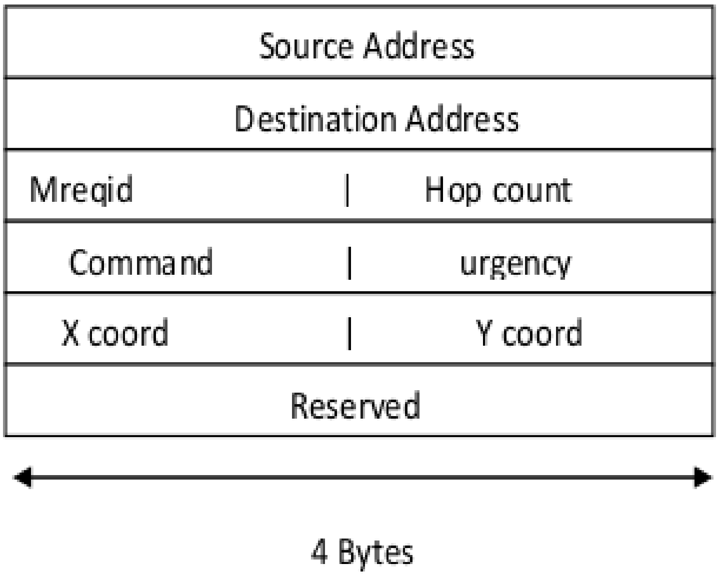

Figure 5.4 Detailed MREQ Packet Structure

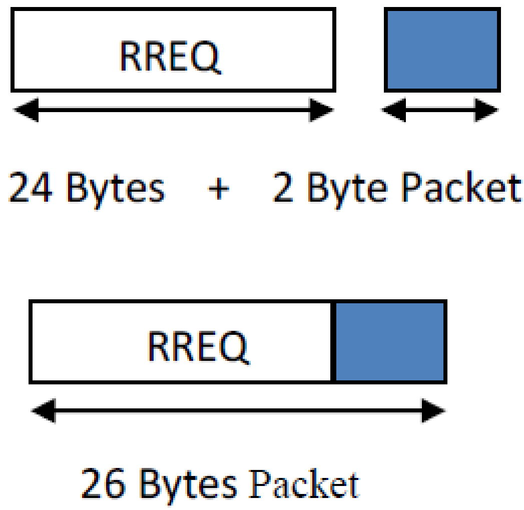

Figure 5.5 RREQ Packet with additional byts for location information

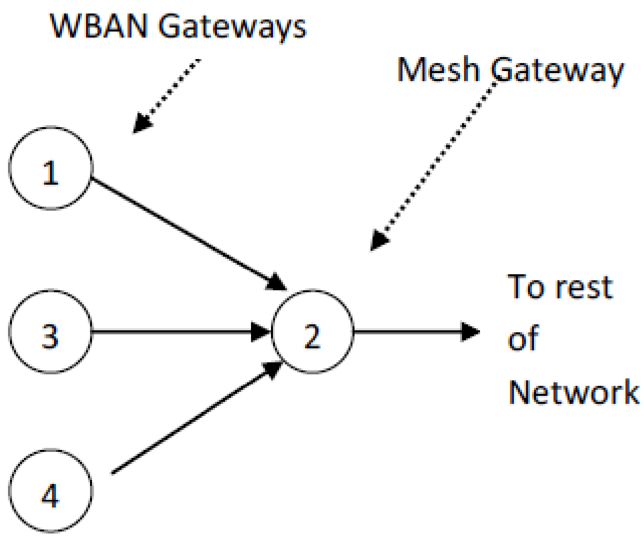

Figure 5.6 Mesh and WBAN gateways

Where :

γ_i = size for each RREQ packet i.

ξ_i = wait for RREP time

$$T_1 = \frac{\gamma N}{\xi N} = \frac{\gamma}{\xi} \tag{5–4}$$

When a path is discovered successfully after reception of a RREP, the next RREQ packet has to wait a total of *Path discovery time + packet transmission time* Thus traffic intensity becomes:

$$T_2 = \frac{\sum_{i=1}^{K} \gamma_i}{\sum_{j=1}^{K} \zeta_j + \sum_{s=1}^{K} \frac{X_s}{R_s}} \tag{5–5}$$

Where :

ζ_j = is the path discovery time.

X_s = is Data packet Size for packet s in the WBAN.

R_s = is the transmission rate for packet s.

Total traffic intensity experienced by the gateway from a WBAN gate way is expressed as,Considering the data packets:

$$T_3 = \frac{\sum_{s=1}^{K} X_s}{\sum_{s=1}^{K} \frac{X_s}{R_s}} \tag{5–6}$$

Where:
X_s is Data packet Size for packet s in the WBAN.
R_s is the transmission rate for packet s.
Total traffic Intensity experience by the mesh gatewway becomes:

$$T_{mgw} = T_1 + T_2 + T_3 \tag{5–7}$$

In equation (5–2),P_B is the blocking probability,given that the mesh gateway node has a finite buffer size of B, that the Buffer in the Mesh gateway is full. Therefore packets which arrive when the buffer is full are dropped. Given two packets from the same node, in the event that a route is not found in the routing table, the Route Request (RREQ) is sent into the network .The interval between two subsequent RREQ packets is the RREQ timer expiry or the Route Reply (RREP) timer.

Therefore if a path has not been discovered, another RREQ is sent until a path is found or the maximum number of retries is reached.Therefore a route request is sent every wait time for RREP expiry and at every path discovery expiry period.There are several reasons why the timeout may expire before the path is discovered. In this work the cause is congestion at the mesh gateway or a node has shutdown due to battery outage. Congestion occurs when the buffer at the gateway node is full,therefore every packet which arrives is dropped, including the AODV control packet , RREQ.

The assumption is that all RREQ are generated successfully and the wait for RREP interval is exponentially distributed with parameter t.It can be observed in (5–3) and (5–4) that for the case where there is no established path to the sink, the equation becomes deterministic. We take that all the AODV packets to be of the same size,24 bytes, therefore the equation becomes.This becomes deterministic. When a path is discovered successfully after reception of a RREP, the next RREQ packet has to wait a total of Path discovery time + packet transmission time.

5.3.2 Network Coding Considering Different Qos Levels In WBAN

WBAN standard IEEE802.15.6 defines 0-7 QoS levels of packets. To use a network coding solution accounting for the different Qos levels,each neighbouring coding node is assumed to have an extra amount of reserved buffer. Given a multihop route, a node will receive indication in the header or in the signal preamble on the type of traffic Qos class for the incoming packet. This signaling information indicates the traffic type about to be received, the time of arrival of the traffic and the delay tolerance.

The incoming packets can find queued or buffered packets in the relay node. Therefore, a certain number of packets in the buffer need to be transfered or cleared from the buffer in order to meet Qos requirement according to the class of the incoming packet. Each packet class has a requirement for the number of cleared packed to maintain the delay requirement. Real time traffic and emergency data have strict and stringent requirements on delay. A buffer clearing factor is defined for each class given by ρ_i for i = (0,1,2,..,7).Given that the number of packets in the buffer at the time of the incoming high priority packet is given by K. i.e $X = (x_1, x_2, x_3, x_4 ... x_K)$ where X represents a packet.The amount of packets that need to be transfered to the sorrounding nodes given the buffer clearing factor is given by equation 5–8.

$$K\rho_i = N_{cp} \tag{5–8}$$

which becomes :

$$K\rho_i - N_{cp} = 0 \tag{5–9}$$

Where N_{cp} represents the number of coded packets that need to be transfered to the sorrounding nodes. The value of the clearing factor , ρ_i , takes on values from 0-100. For example highest priority ECG may take a value of 100 or a factor of one signifying that all packets in the buffer should be cleared.Equations 5–10 and 5–11 thus represent the total the throughput or data rate needed to transfer packets in the buffer to the sorrounding node buffers in order to meet the requirements of the stringent delay requirements of high priority packets.

$$Thr = \frac{\rho_i N_B}{T_c + \frac{\rho_i N_B}{\mu_c}} \tag{5–10}$$

$$Thr = \frac{1}{\frac{T_c}{\rho_i N_B} + \frac{1}{\mu_c}} \tag{5–11}$$

$$R\Delta T = N_{cp} \tag{5–12}$$

Where T_c is the time taken for combining packets in the buffer,N_B is the number of packets in the buffer which is the same as K indicated in equation

5–8.μ_c is the service rate for the coded packets.Considering the different Qos level of traffic in WBAN, the problem becomes finding a code such that (5–12) is satisfied. The coding function is such that it increases the clearing rate of the buffered packets depending on the Qos level of the incoming packet selected by the clearing factor, ρ_i .

5.4 Results and Discussion

The results of the numerical evaluation show the traffic intensity as observed from the mesh gateways point of view.Fig. 5.13 shows the traffic intensity on the mesh gateway due to the route request, RREQ when the wait timer for RREP elapses. The timers in the evaluation have been assumed to have an exponential distribution. For the RREQ, which are control packets, the traffic intensity is in the order of Kilobytes, which is manageable by a single Mesh gateway.However, this can quickly rise depending on the number of WBAN connecting at the same time.

Fig 5.7 shows the traffic intensity by the RREQ with an interval of the path discovery timed.Fig. 5.14 shows that when data packets are considered, as well, the intensity quickly rises depending on the traffic type. This has implications on the buffer and energy of nodes. Introducing of a new packet structure and attaching additional fields for data seem to have no such effect on the gateway.

Various throughputs are generated ,given that the route reply timer is exponentially distributed, the route request packet in AODV has a size of 24 bytes.Thus an rreq packet is sent when the route reply timesout.In this case we use 100 packets which are sent.100 packets are sent and each packet times out once before the routeroute reply becomes successful.Therefore a route request RREQ is sent every RREP timer interval,giving a throughput of:

$$Transmissionrate = Sizeof(RREQ)/RREPtimer \tag{5–13}$$

The cummulative value of the throughput is then plotted.Equation 5–13 is the transmission rate that is obtained given that a control packet ,in this case a Route Request (RREQ) is sent every RREP timer expiry.This means the shorter the RREP timer expiry time,the more packets or intensity the gateway will experience.Since RREQ is relayed to the other nodes in the network,shorter RREP timer can quickly overwhelm nodes which have limited resources in terms of buffer size.Once the buffer is full due to control packets,this means that other data packets from other nodes are denied access to the node resulting in packets being lost.Various traffic intensities are plotted ,and in AODV,the expiry timer is a deterministic value,however in this secenario,it is modelled to be a random number which is exponentially distributed with various rate parameters given as 0.01 , 0.05 , 0.09 , 0.1 and 2.

In Fig. 5.9,the various traffic load on the gateway nodes are compared rate parameters.From the graph,the trend that is observed the load on the gateway is higher with a smaller rate parameter and reduces as the rate parameter is increased to 2.In this scenario,the traffic intensity is due to the sent route requests from different nodes attached or connected to the gateway.

The graph shown in Fig. 5.12 shows another scenario where the traffic intensity on the gateway due to various rate parameters is compared.The traffic load on the gateway is due to RREQ given a succesful path discovery time and the time to send a variable lenth data packet. The graph in Fig. 5.14 shows the comparison of the traffic intensity on the gateway at various rate parameters and given that time is exponentially distributed.The graph shows a comparison of the traffic intensity on the gateway node due to a combination of the intensity due to RREQ given pathdiscovery timer,rrep timer and the transmission time or service time of the data packets through the mesh gateway node into the multi-hop mesh network.When the traffic intensity is combined at various rate parameters of the exponential distribution of timers,the traffic intensity is almost the same value at all the rate parameters evaluated.

For the scenario of incoming packets with strict and stringent Qos as discussed previously in the previous section, Figs. 5.15 - 5.18 shows the comparison of various service rates and the resulting data rate that is needed in order to meet the buffer clearing factor requirements.The buffer clearing factor is evaluated at 10,40,80 and 100 percent with different service rates for the coded packets at 1250, 12500, 25000 , 75000 and 125000 bytes per second.As expected for the same amount of time needed to combine packets in the buffer to form network coded packets,higher service rates provide a higher throughput and therefore much faster to transfer packets to the sorrounding nodes than the low service rates at all the different buffer clearing factor values.

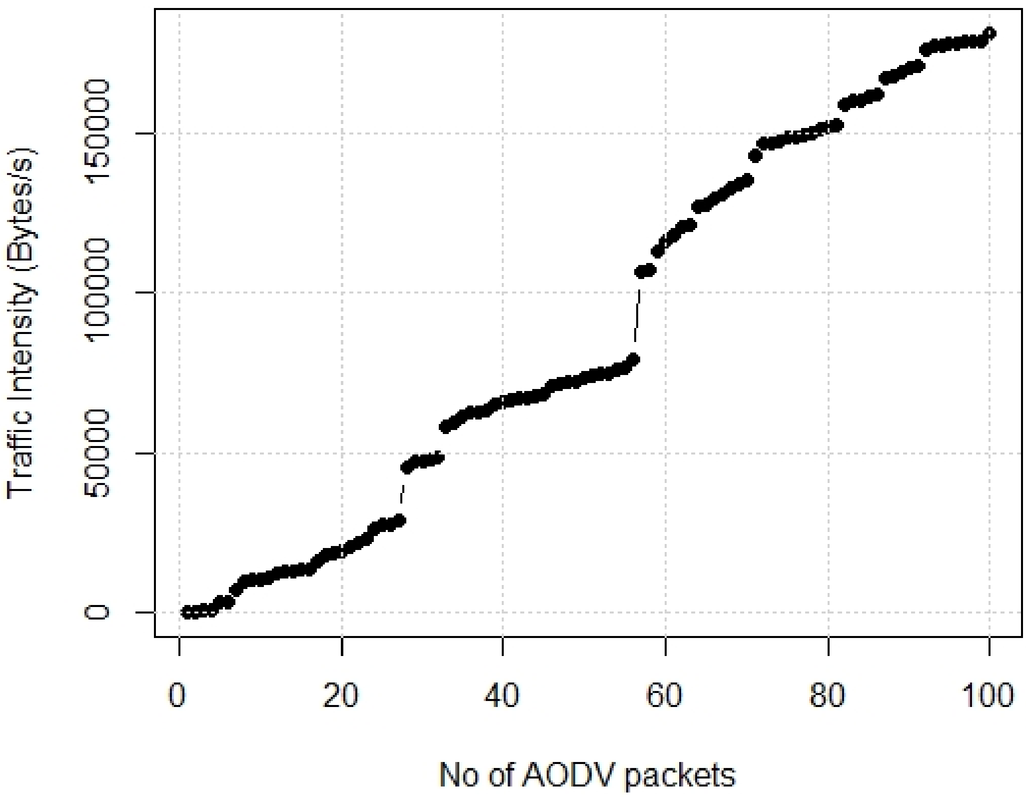

Figure 5.7 Traffic Intensity due to RREQ after successful route discovery

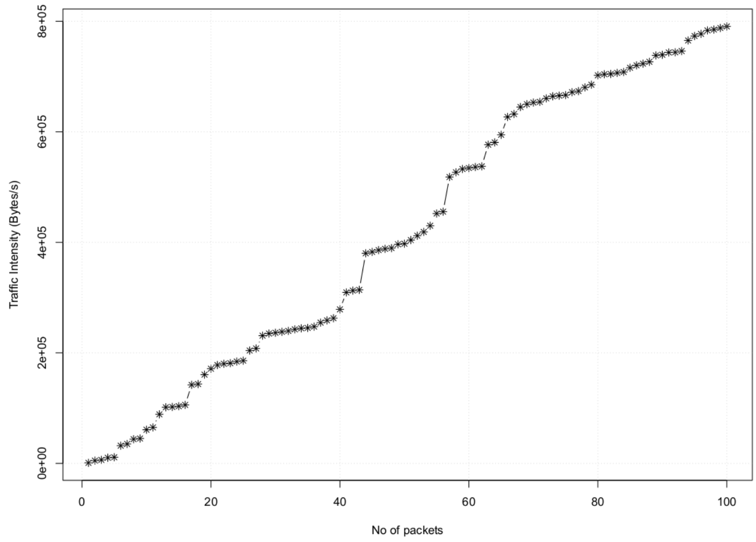

Figure 5.8 Traffic Intensity due to RREQ

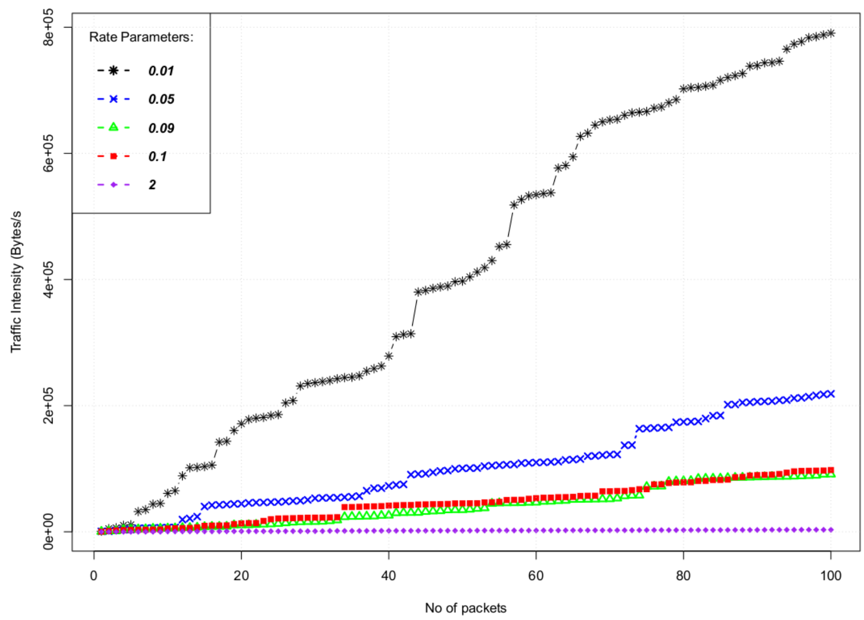

Figure 5.9 Traffic Intensity due to RREQ successful route discovery for various rate parameters 0.01,0.05,0.09,0.1 and 2 of the route reply timer exponentially distributed at

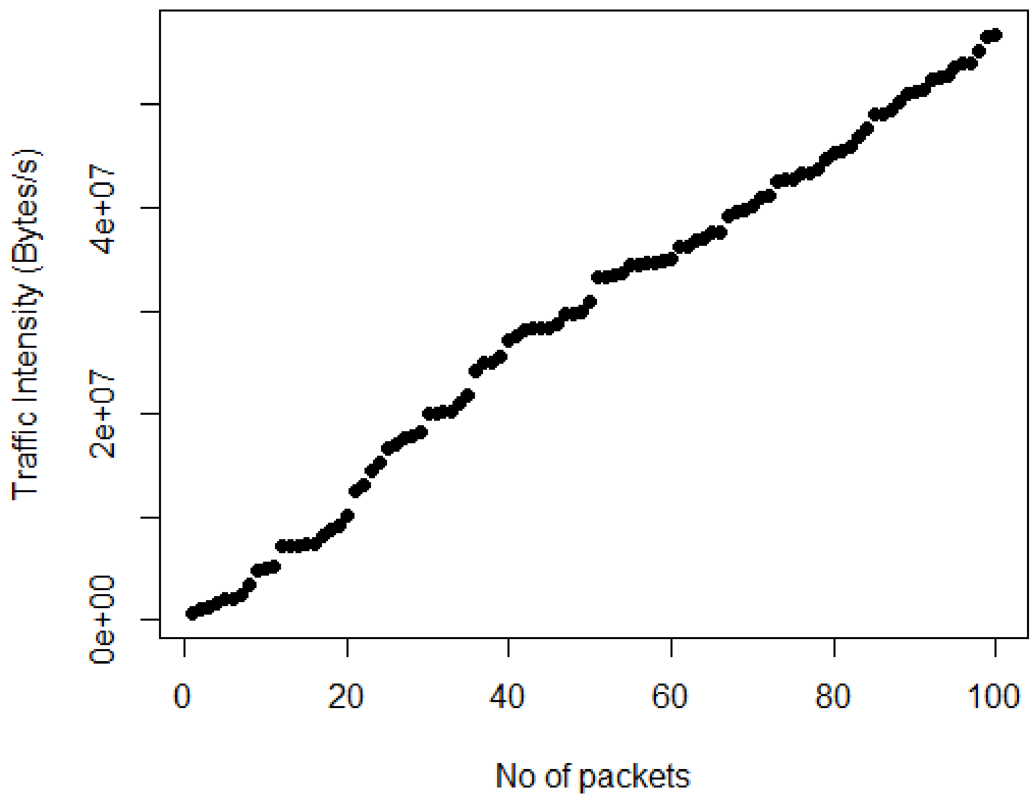

Figure 5.10 Traffic Intensity due to combined RREQ and data Packets

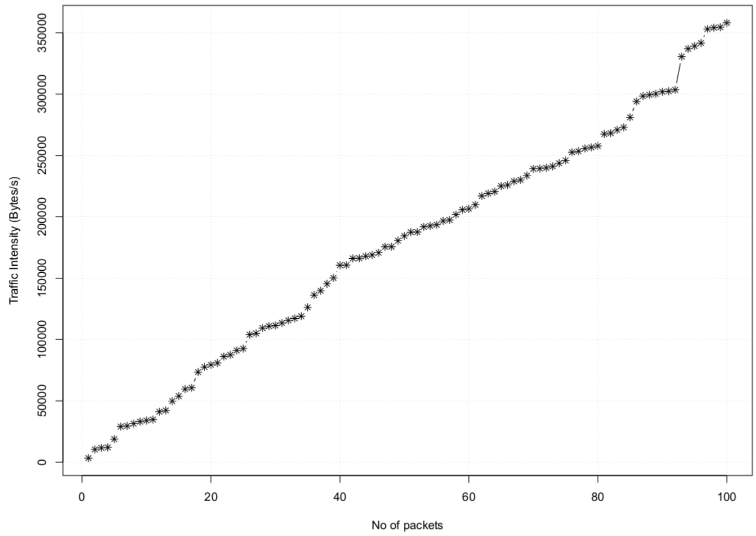

Figure 5.11 Traffic Intensity due to RREQ after successful route discovery at 0.01 rate parameter

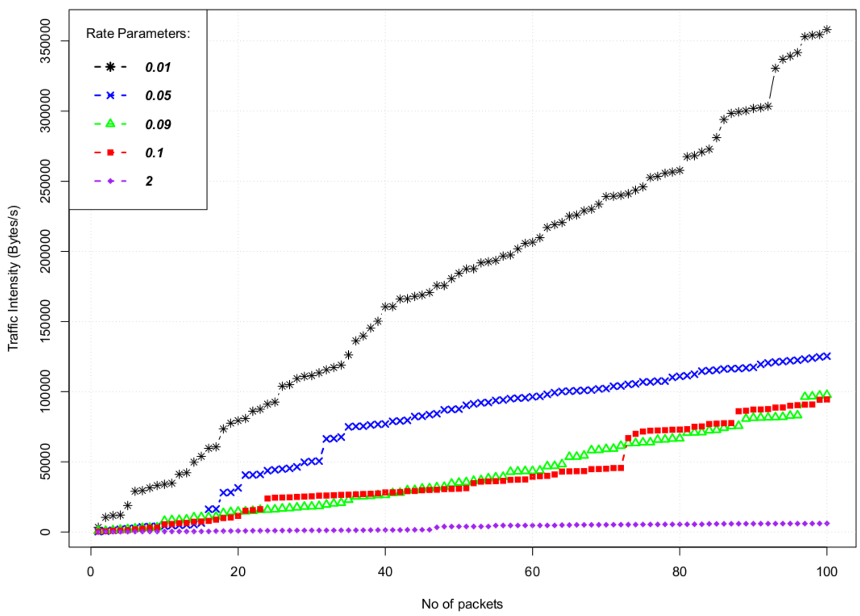

Figure 5.12 Traffic Intensity due to RREQ and DATA for various rate parameters 0.01,0.05,0.09,0.1 and 2.

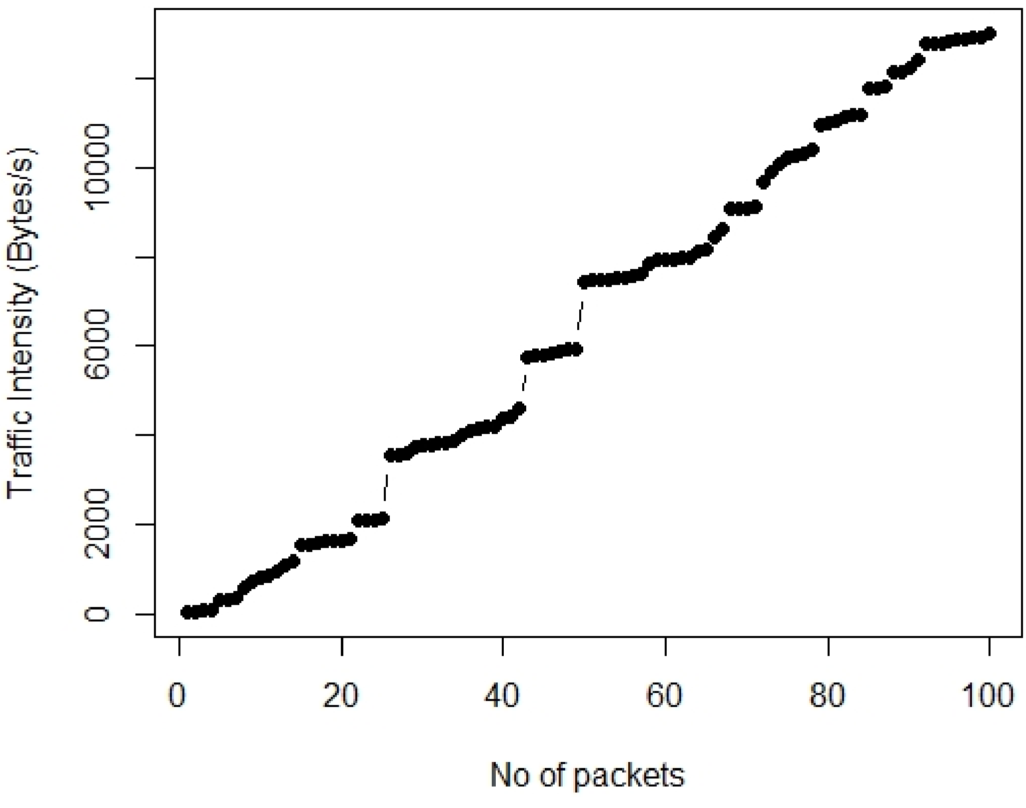

Figure 5.13 Traffic Intensity when the RREP is not received on time

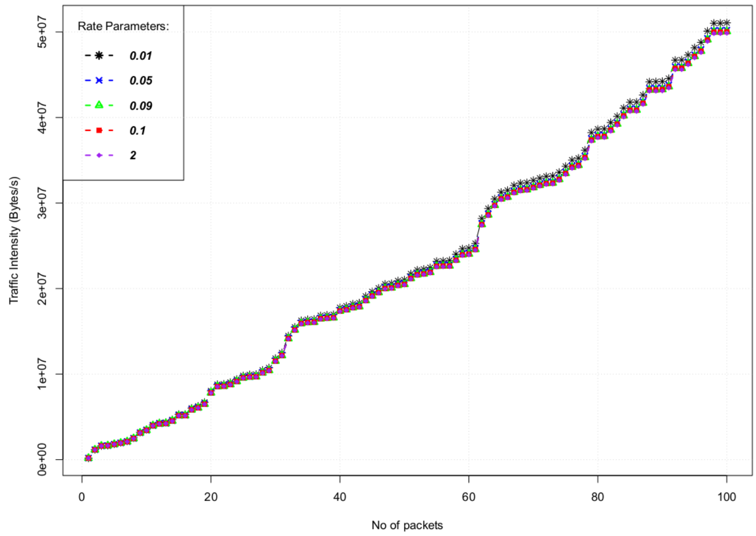

Figure 5.14 Combined traffic intensity at various rate parameters

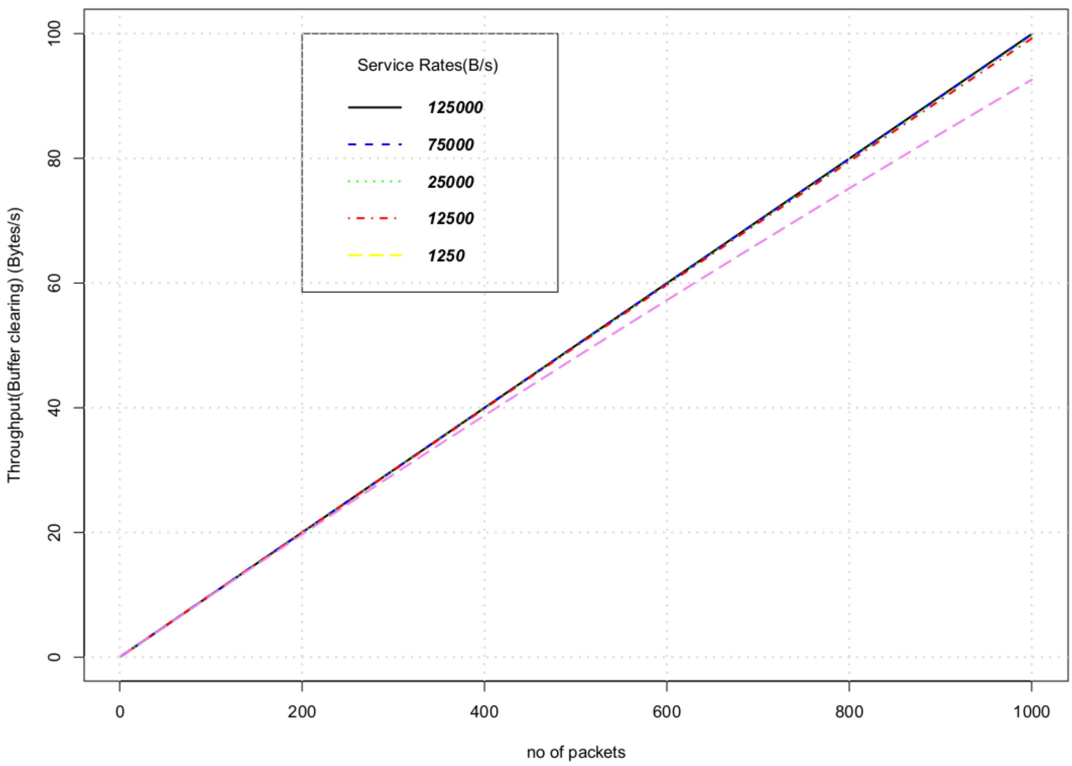

Figure 5.15 Throughput with buffer clearance factor set to 10% with various service rates.

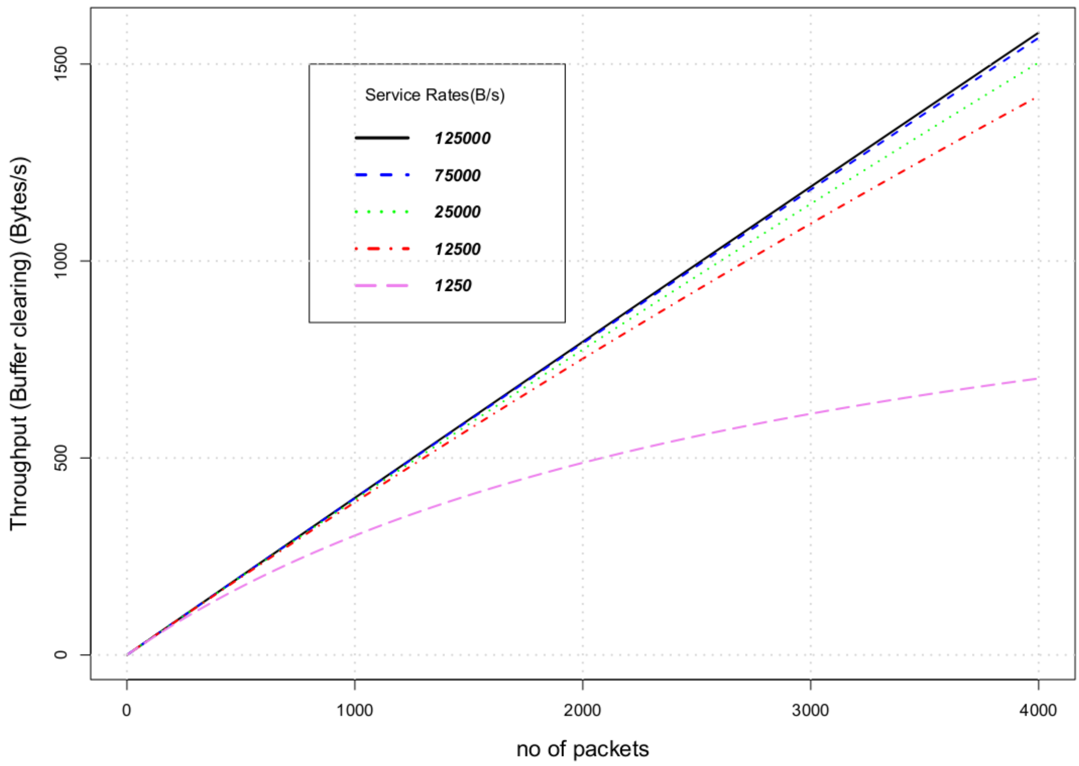

Figure 5.16 Throughput with buffer clearance factor set to 40% with various service rates.

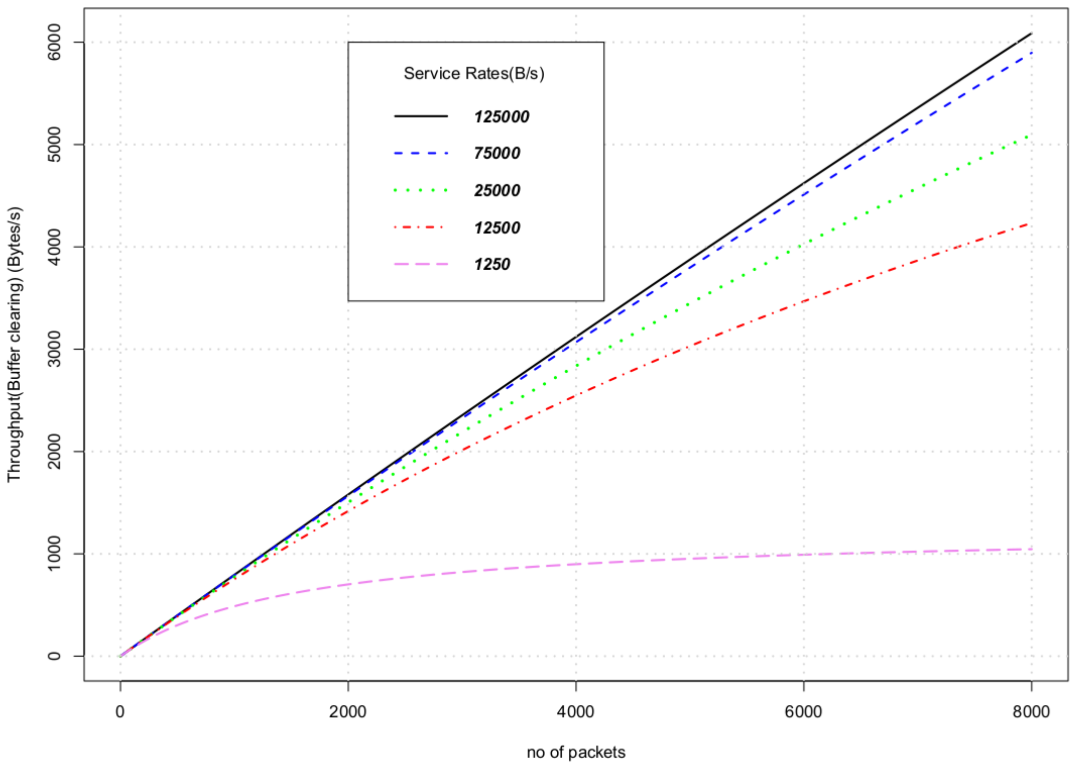

Figure 5.17 Throughput with clearance factor set to 80%.

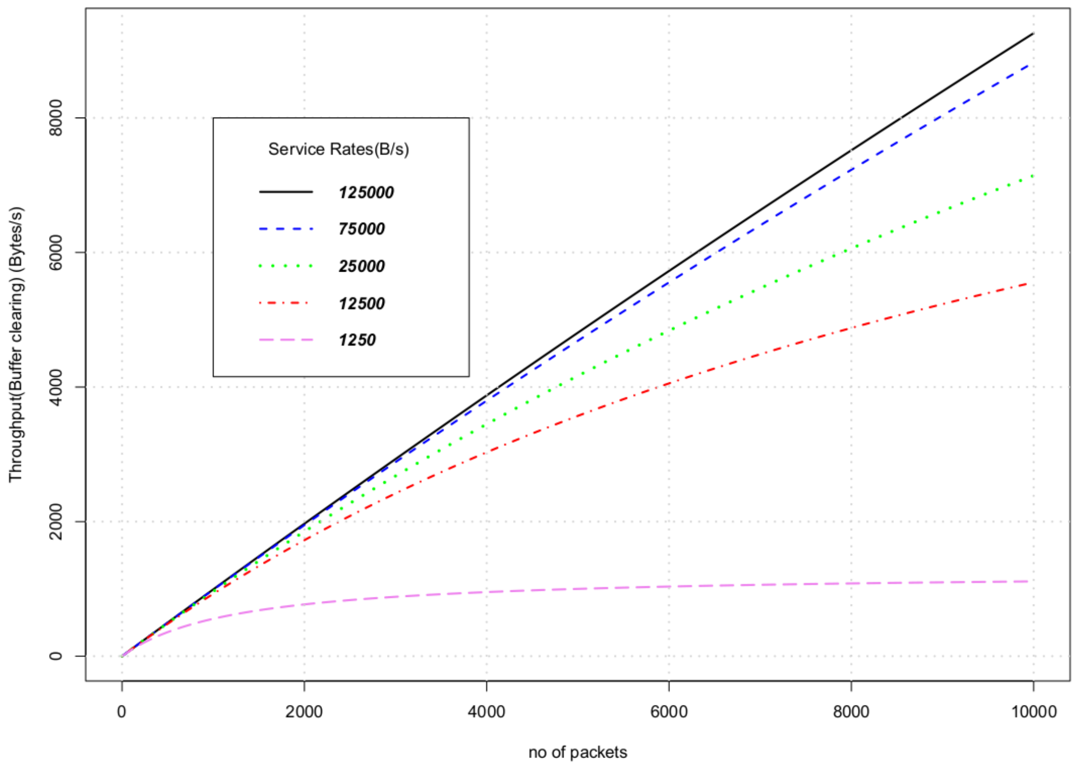

Figure 5.18 Throughput with clearance factor set to 100%

5.4.1 Drawback

The drawback of this method is the complexity that is introduced is introduced for obtaining location information. The coding node can obtain location information based on Array antenna techniques or by sending packets embedding location information in the packets they send. Both methods need to be evaluated for delay and throughput performance, and power requirements depending on the signaling load .This complexity can be the difference in implementation in networks within constrained environments such as internet of things (IoT).

5.5 Conclusion of Chapter

This chapter looked at a method to implement a protocol for Network discovery of potential network coding nodes. By utilizing angle of arrival information and location information embedded in the packets header,potential coding nodes are identified using packets to be combined. The use of location information for each node and how it is obtained has an impact on the complexity of implementation especially in resource constrained environment.Future network designs have to consider the scalability of the network as more nodes increase in the network.

Various network nodes for example Wireless Body Area Network (WBAN) can be interconnected to a mesh gateway which receives traffic from the WBANs to the rest of the multihop mesh network.If routing protocols are used for example AODV by these gateways for route discovery,these may have an additional traffic load on the mesh gateway which can affect the buffer size and power consumption of the gateway node.

Depending on the Qos of the incoming packet,various sizes of packets which are in the buffer can be transferred to the virtual buffers of the sorrounding nodes at various transmission speeds or service rates in order to meet QOS requirements,for example for packets to meet a delay deadline.

Chapter 6

Buffer Extension in Multi-Hop Networks for Packet Delay Improvement

6.1 Introduction

With the increase in the demand for connected devices, more and more devices are connected to the internet. This can be attributed mainly to advances in embedded systems .With the coming of the Internet of things (IoT), it is expected that the number of connected devices will increase to about 50 Billion by 2020. This chapter introduces theoretical concept of the notion of buffer extention.The network structure is a two tier network consisting of access points and their client hosts. The access points form the first tier which forms a multihop relay network from one access point to the other. The access points are known by several names such as servers, or gateways.

The client nodes connects to the access point randomly,therefore the access point or relay node handles traffic from within its own cell as well as traffic from the other access points which it relays. Assuming the gateway has a finite buffer and there are two classes of traffic, High priority and ordinary class packets,in heavy traffic conditions, packet losses can be experienced due to queueing delay for the higher class of packets. By using discrete event simulation, end to end delay in this architecture is observed as well as the changes in the buffer length. If no collision is experienced ,packets will contend for buffer space.Through simulation observation of the buffer of the source access point for high priority packets, the buffer size for the relaying node and the end to end delay specifically for the high class packets is noted. What is observed, at low node count, relay node buffer is stable and there is very little delay as would be expected by intuition.The scheme being investigated to improve on delay is similar to an implementation of a priority queue, however in this case ,when the buffer has reached its maximum value and a priority packet arrives, instead of the high class packet queueing, hence being delayed, the packets give way, instead of the packets being discarded, they are transferred to the neighboring nodes. What is observed is that the buffer size has a lot of dynamics in large networks and a change in traffic characteristics ripples through the multi hops.

6.2 Distributed Buffer Extension

In order to address the problem described above, a scheme of distributed buffer extension at a relay node in a multihop network is investigated. Firstly, incoming packets of high priority to the relay node have stringent deadlines. Secondly they maybe highly intolerant to packet loss. The problem of packet loss is addressed by evaluating how much throughput is needed to send distribute packets of lower priority to be stored in the neighboring nodes. Extension of the buffer is done virtually based on queueing theory from the probability that there are N packets in an M/M/1/B queue.The first M stands for Markovian(Poisson) process which is the arrival process,the second M is for interdeparture time (service time) and exponentially distributed.The number of servers is 1 and B is the buffer size.The buffer is as shown in Fig. 6.1.

Given that the number of surrounding nodes is D and the amount of reserved buffer by each node be ϕ. Buffer extension due to the surrounding node will depend on the density and the probability of finding the sorrounding nodes buffer is full. This has the effect of reducing the blocking probability and hence the packet loss. Whereas probability of Packet loss is improved by the virtual buffer extension, coding the incoming packet flow or the packets in the buffer gives a reduction in the expected latency and reduction in packet loss. There are several parameters which are considered when looking at quality of Service (QOS).These include data rate, bit error rate, packet error rate, duty cycle, desired battery lifetime and delay. These parameters all depends on the service being considered.

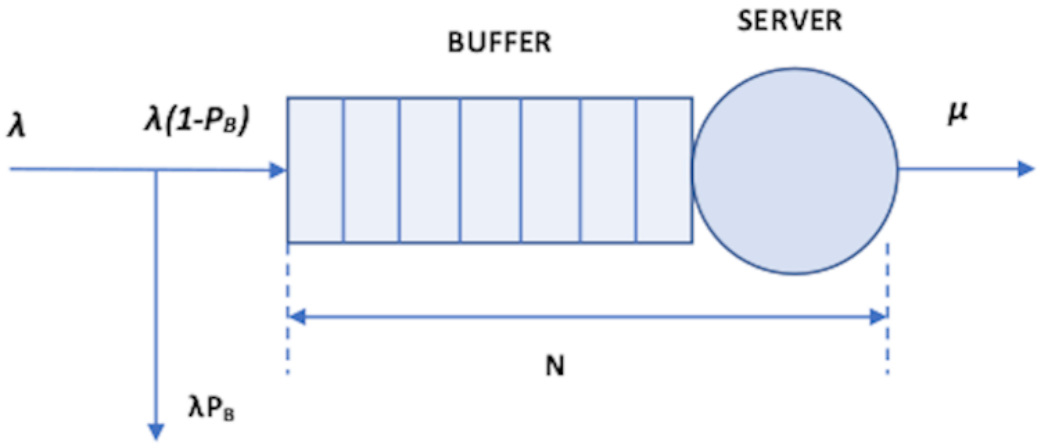

Figure 6.1 Buffer Model

6.2.1 Buffer Clearing

Packets in the buffer are cleared according to the class of traffic.Each class of traffic has its own requirement for the number of packets to be cleared to maintain the delay requirement. Real time traffic and emergency data have stringent delay requirements.Using a buffer clearing factor or scale parameter for each QoS class, the value of the clearing factor takes on values from 0-100 percent .For example highest priority may take a value of 100 percent or 1 as the clearing factor signifying that all packets in the buffer should be cleared if an incoming packet of higher priority enters the buffer at the relay node.For emergency packets or highest priority of packets, when the buffer is full, all the contents of the buffers should cleared or offloaded to the neighboring nodes.In real networks, to address the problem of QoS and congestion, several strategies are employed, including Trunk reservation and Virtual channel protection.

The flow chart in Fig 6.5 show how the Buffer clearing algorithm is implemented. The preamble of the incoming packet is first read, priority information can be encoded in the preamble. If the Packet is not a high priority packet and the buffer is full, it is dropped,otherwise inserted into the transmission buffer waiting for transmmission. If however the incoming packet is high a priority with respect to the packets in the Buffer, The packet size is calculated. If the

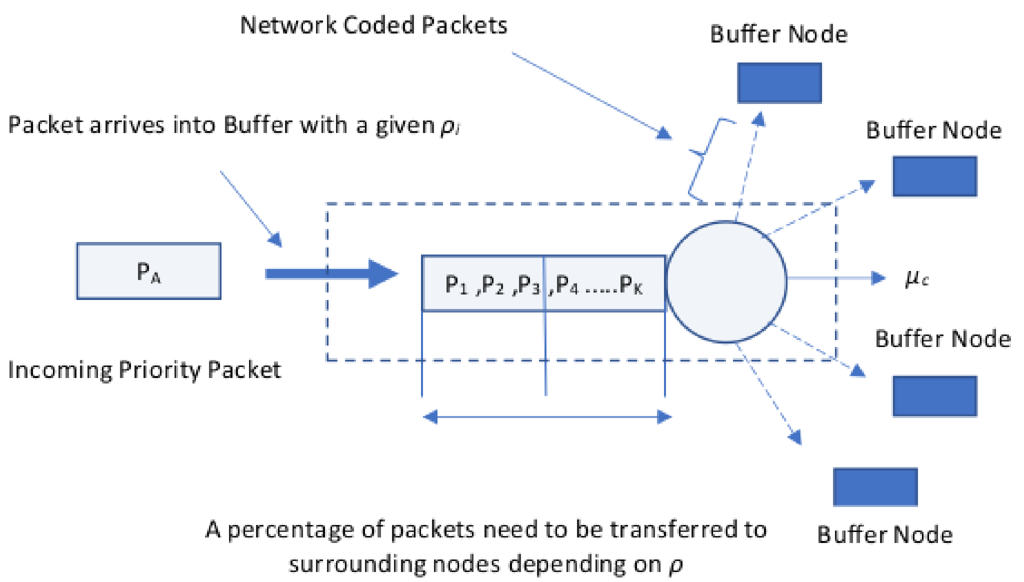

Figure 6.2 Packet Transfer to Sourrounding Neighbour Nodes by Relay Node

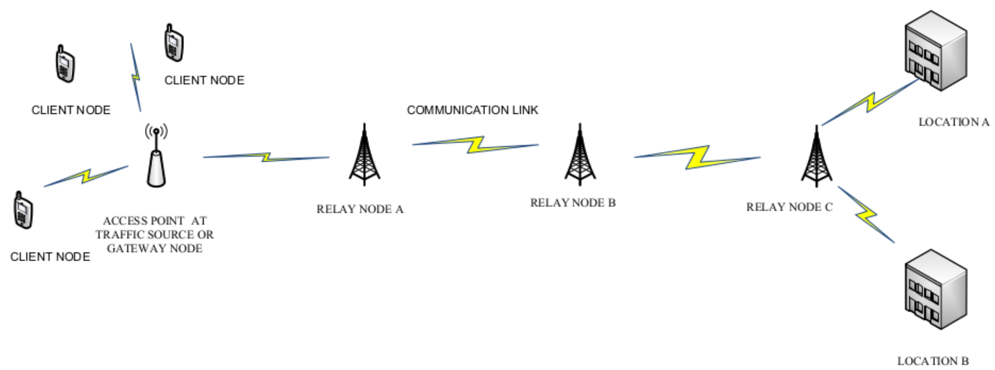

Figure 6.3 Multihop Network with no sorrounding nodes or client on the intermediate nodes.

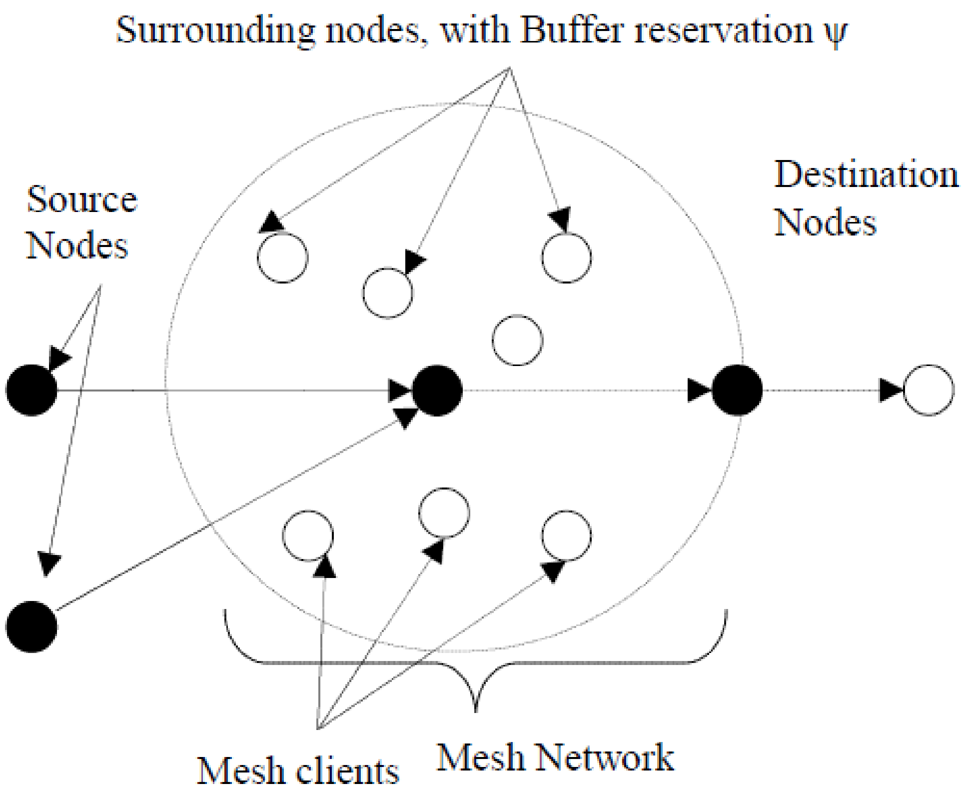

Figure 6.4 part of a mesh network with shaded nodes as main nodes and the non shaded nodes as the client or sourrounding nodes.

packet cannot fit into the buffer, the node density is checked. The node density ensures there are enough surrounding nodes such that packets can be transferred to the surrounding nodes, then the process is complete. If there is an alternative path, from the surrounding node, the packets are delivered to their destination. If the node density is less than a threshold, then the high priority packet is discarded.Fig6.5 illustrates the flow chart for buffer clearing in order to improve on the delay.

Fig.6.7 shows the result of implementing a buffer clearing scheme to improve on delay. In this instance the buffer size limit was 256 packets, and once a high class packet arrives, all the packets in front are cleared to give way. The gaps in the graphs are the instances when the buffer is cleared and also when a high priority packet arrives. The queues quickly builds up because of the high traffic intensity.

6.3 Application to IoT

In the case of IoT with a LoRa radio, to clear the buffer for the incoming data ,part of the preamble length assuming perfect synchronization of 5 symbols is used. Meaning the data in the buffer has to be cleared within the remaining preamble time.At which point the receiver will synchronise with the preamble can not be know for sure , assuming perfect synchronization at the receiver of the incoming preamble, in case of Lora, there are 10 preamble symbols [25]. It takes 5 symbols to synchronise with the preamble. The remaining 5 symbols is the time in which to clear the buffer in the node. The buffer length for most IoT devices is in hundreds of bytes for most devices on the market. In the case of 256 Bytes buffer and its full,all the 256 packets have to be cleared within the remaining time after synchronization. A LORA packet may arrive at a node using one spreading factor and leave on another spreading factor to enable a higher data rate than incoming or current one at the node.This will be discussed in **chapter 6.**

6.4 Results and Analysis

It can be observed that, at low traffic levels, the buffer fluctuations are low. In this way the high priority packets experience very low latency. As the traffic

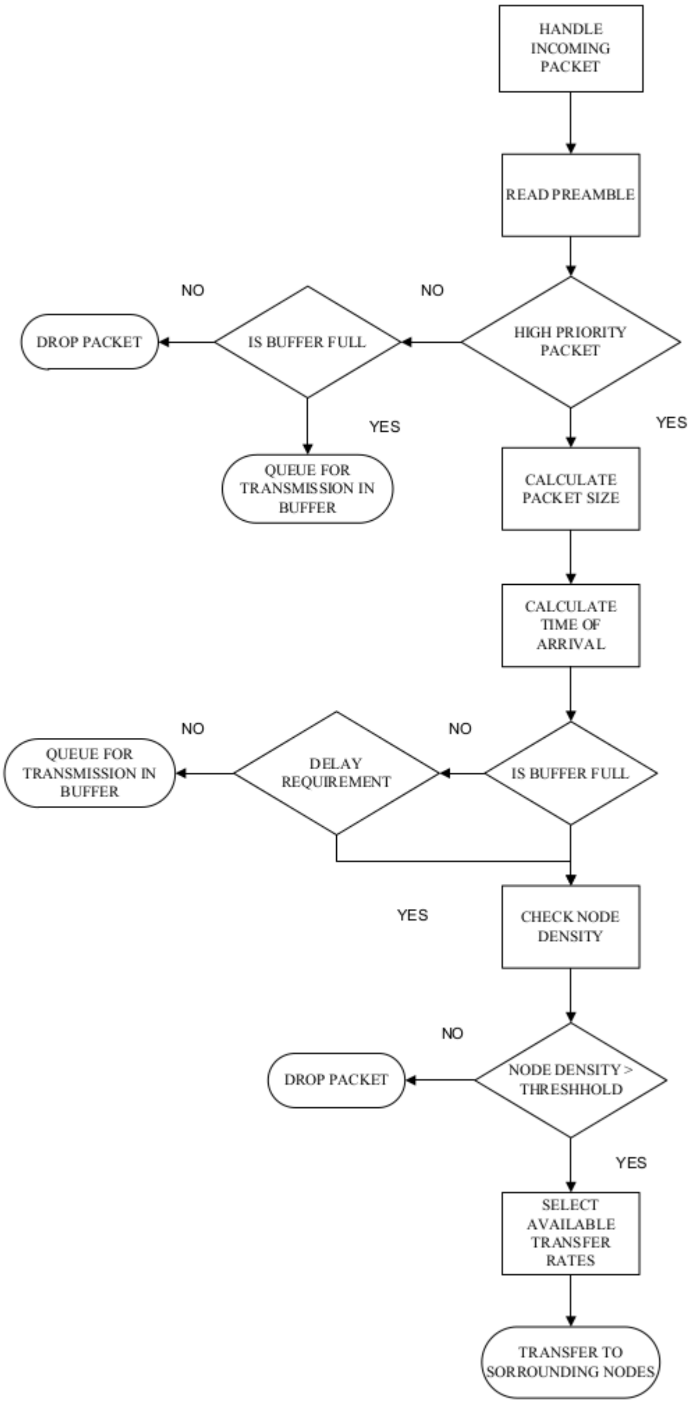

Figure 6.5 Buffer Extension Algorithm Flow Chart

Table 6.1 Simulation Parameters

Parameter	Value
Channel	Ideal channel conditions
Access Technique	Aloha
Number of Gateway/Base stations	3
Number of nodes	$10,20\cdots 100$
Node distribution	Uniform (0m,1000m)
Simulation Area	1000 m by 1000 m
Packet inter Arrival times	Exponential with mean 1
no of Gateway	3
Gateway 1	(x-150m,y-840m)
Gateway 2	(x-500m,y-500m)
Gateway 3	(x-900m,y-250m)

is increased, the relay nodes experience an increase in traffic which causes the rapid fluctuations in the buffer size.

These rapid fluctuations in turn affect the end to end delay resulting in jitter. Various topologies need to be tried out as well and evaluate the buffer behaviour. Three things buffer were observed; behavior of the relay node/gateway , buffer behavior with time of the source node and the end to end delay for the high priority packets. 50 nodes where considered with a packet size of 160 bits and a transmission rate of 9.375kbps. The observation was under high traffic with an exponential distribution with interarrival mean of 1s. The assumptions are that packets are not lost due to contention and are received without errors, the access method is pure Aloha and nodes are randomly distributed with a uniform distribution in a 1000m X 1000m area. The data rate of 9.375 kbps corresponds to LoRa SF 6 nominal data rate. Figures 6.12, and 6.13 shows the observation for 50 nodes.Fig. 6.6 below shows the structure of the simulation network used and the parameters can be obtained from table 6.1. Figures 6.7 , 6.8 , 6.11 and 6.10 and 6.14 show evaluation at 100 nodes. Fig. 6.15 shows the comparison of the result when the algorithm implemented using the flow chart above for end to end delay of high priority packets. Without the algorithm ,it is seen that the end to end delay rises linearly with as the simulation time progreses.In contrast when virtual extension of finite buffer in the relay node is implemented,the end to end delay is relatively small below 1s.

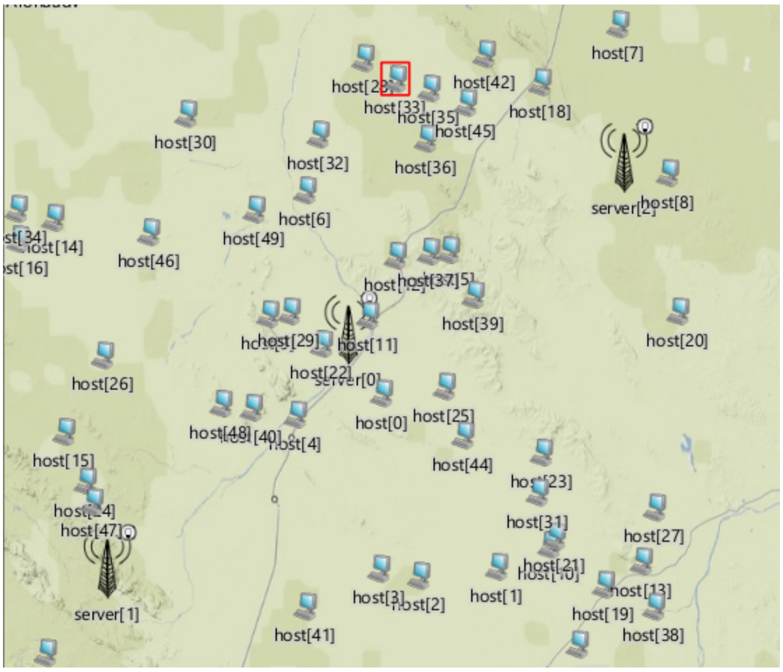

Figure 6.6 Network Simulation

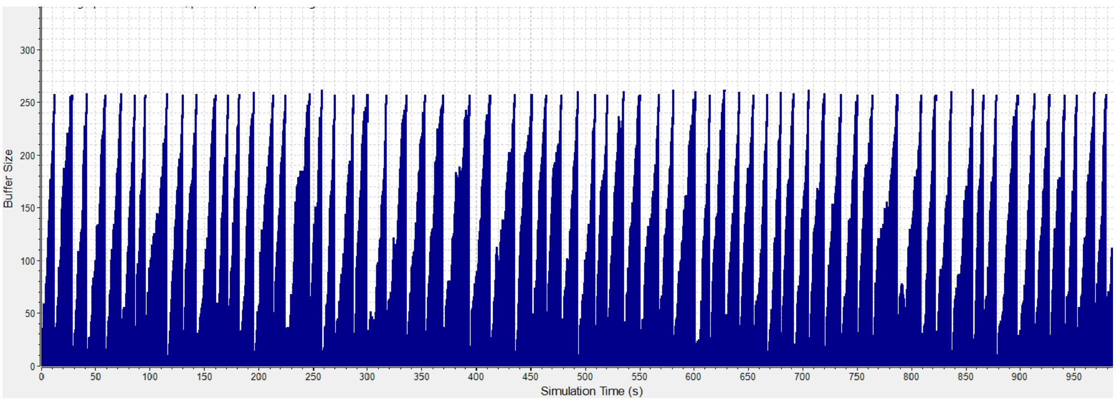

Figure 6.7 Buffer Size Vs Simulation time for 100 Nodes with packet size of 160 Bits and Data Rate of 9.375Kbps.

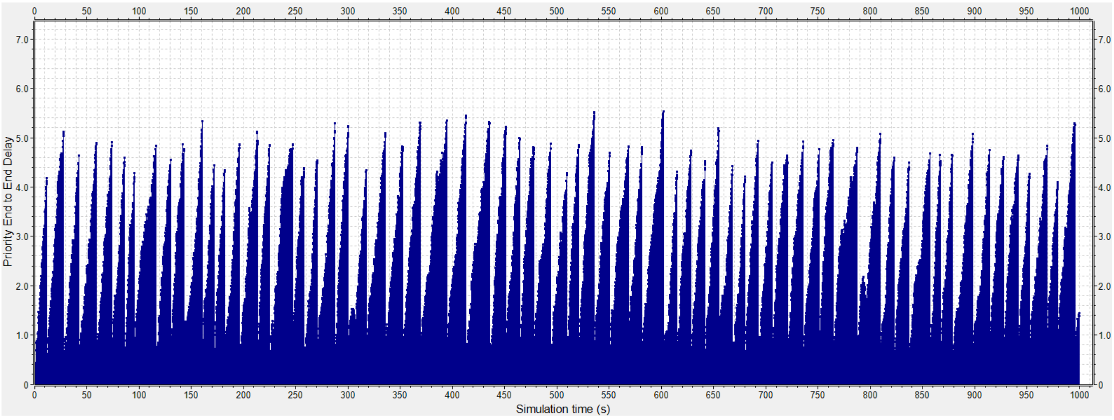

Figure 6.8 Priority Packet Delay Vs Simulation time with 100 nodes,Packet size 160 Bits and data rate 9.375Kbps

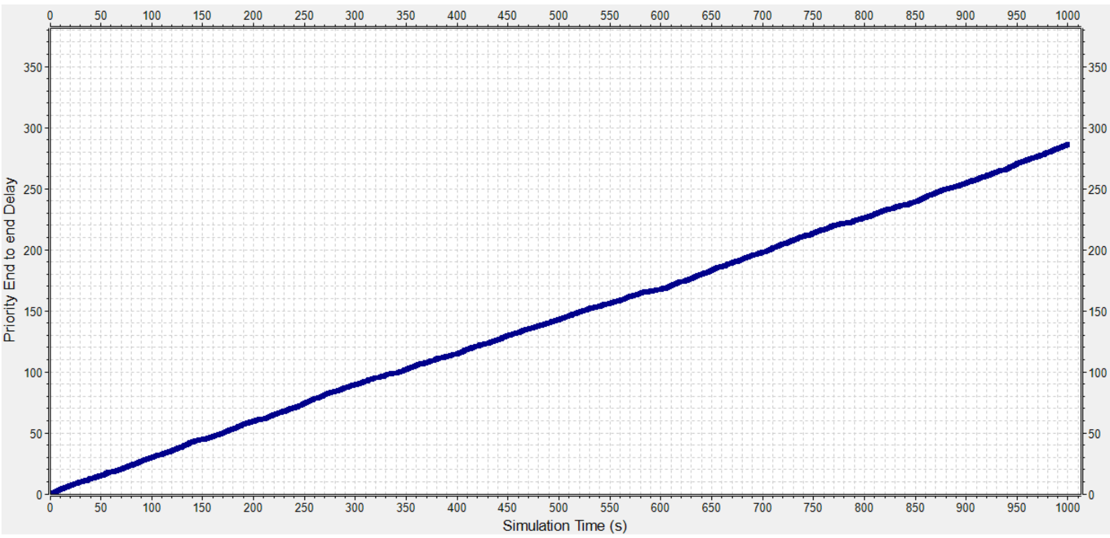

Figure 6.9 Priority Packet Delay Vs Simulation time for 100 Nodes with Packet Size of 160 bits and data rate of 9.375Kbps at destination node

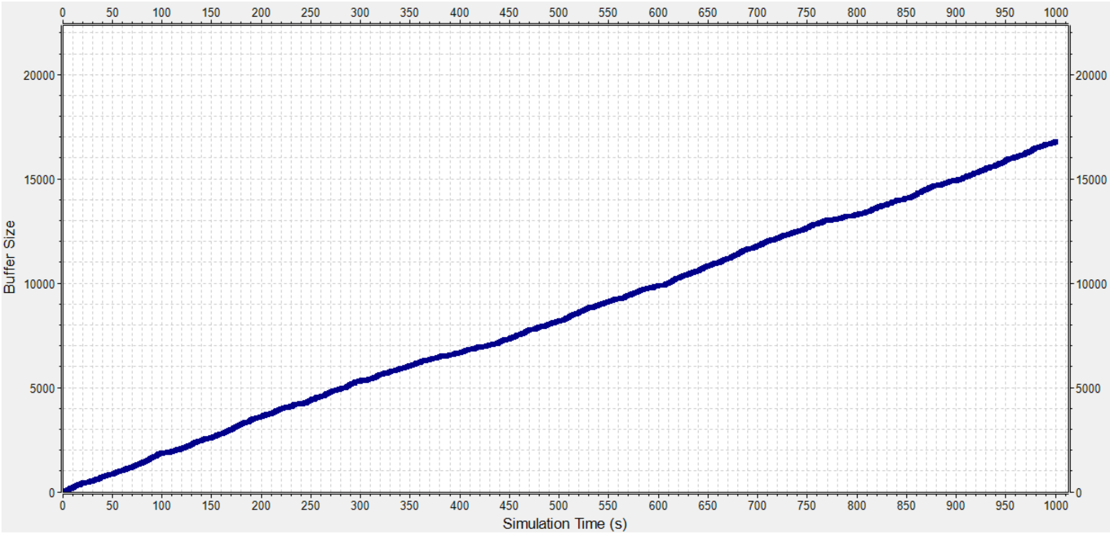

Figure 6.10 Buffer Size vs Simulation time for 100 Nodes with Packet Size of 160 bits and Data rate of 9.375Kbps at Relay node.

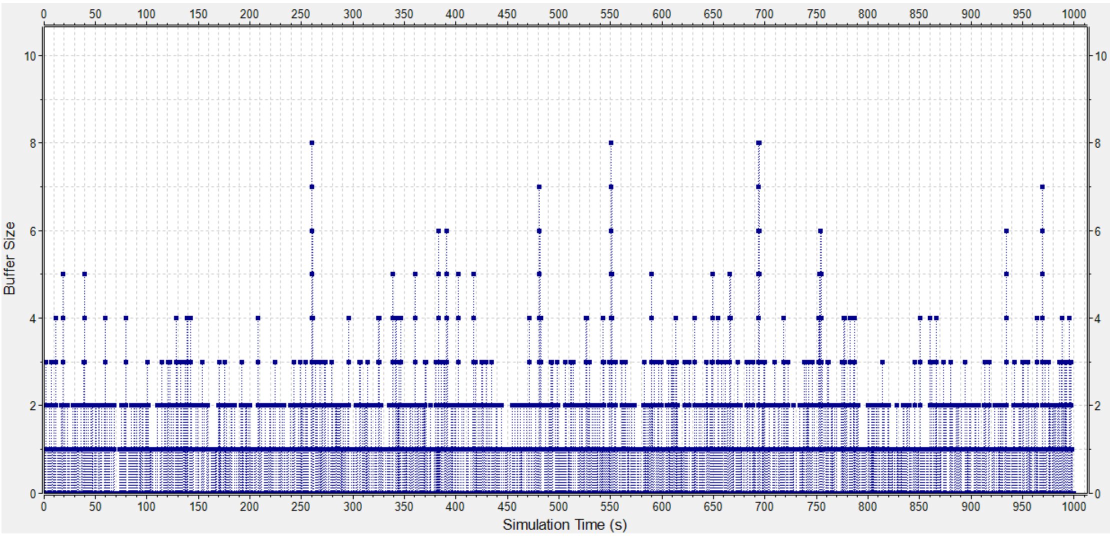

Figure 6.11 Priority Packet Delay vs Simulation time for 100 Nodes,Packet size of 160 bits and data rate of 9.375 Kbps at Source node.

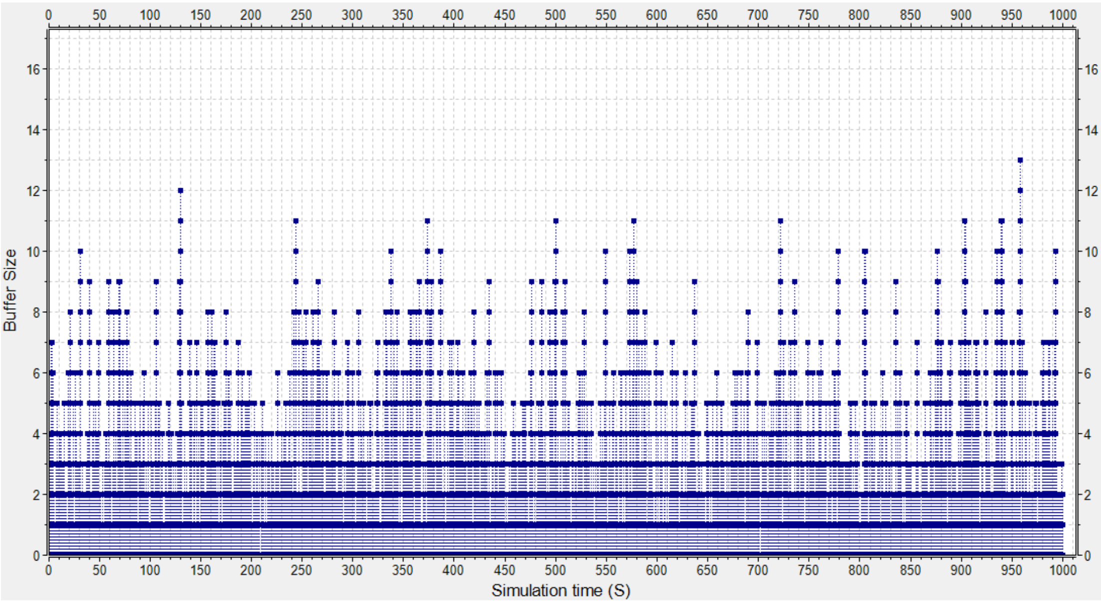

Figure 6.12 Buffer Size vs Simulation time for 50 nodes,Packet size 160 bits and data rate 9.375Kbps at Relay node.

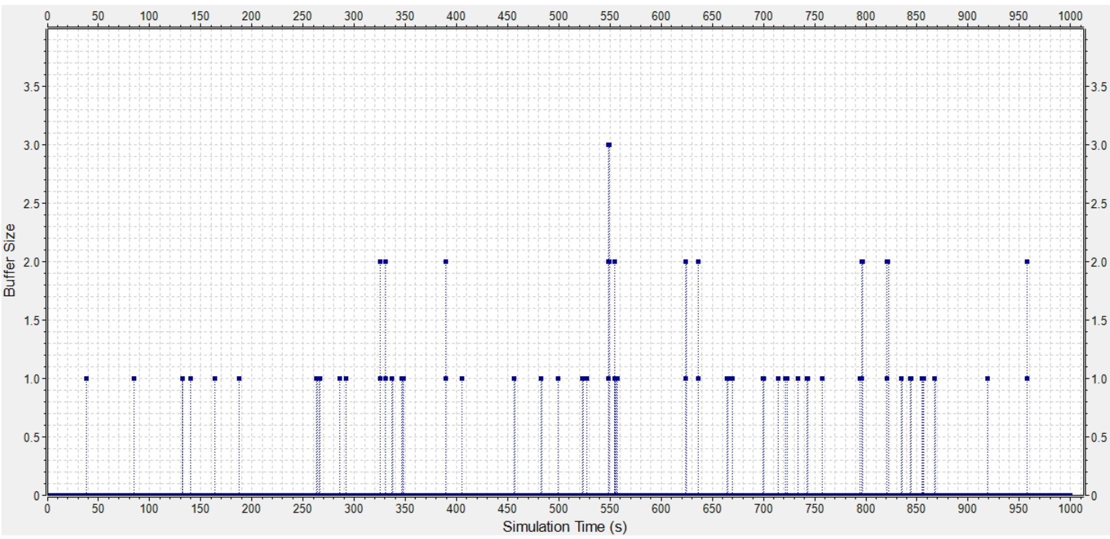

Figure 6.13 Buffer Size Vs Simulation time for 50 Nodes,Packet size of 160 bits and Data rate of 9.375Kbps at Source node.

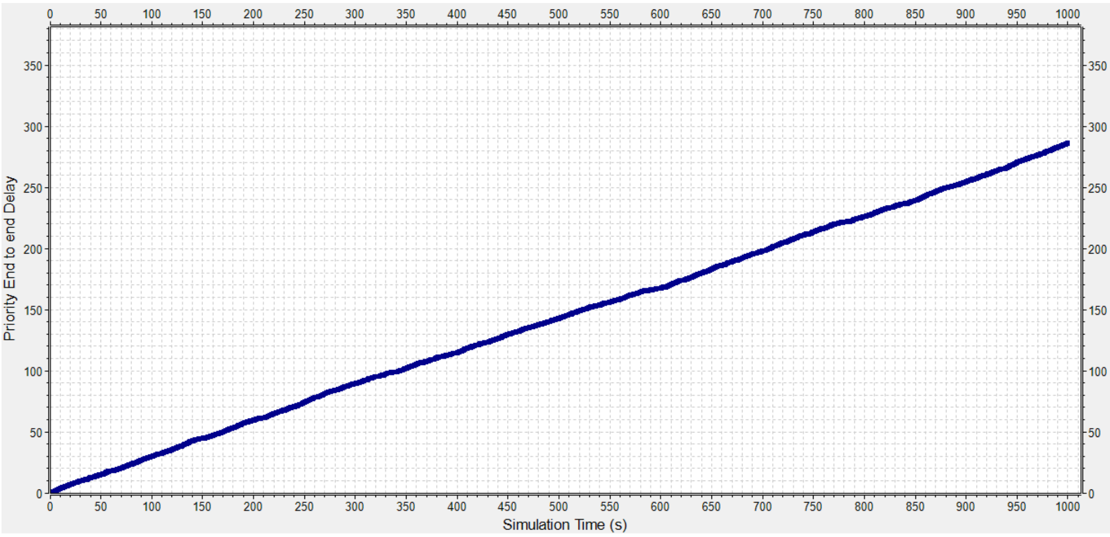

Figure 6.14 Priority Packet Delay Vs Simulation time for 100 Nodes ,160 bits and Data rate of 9.375Kbps at Destination node.

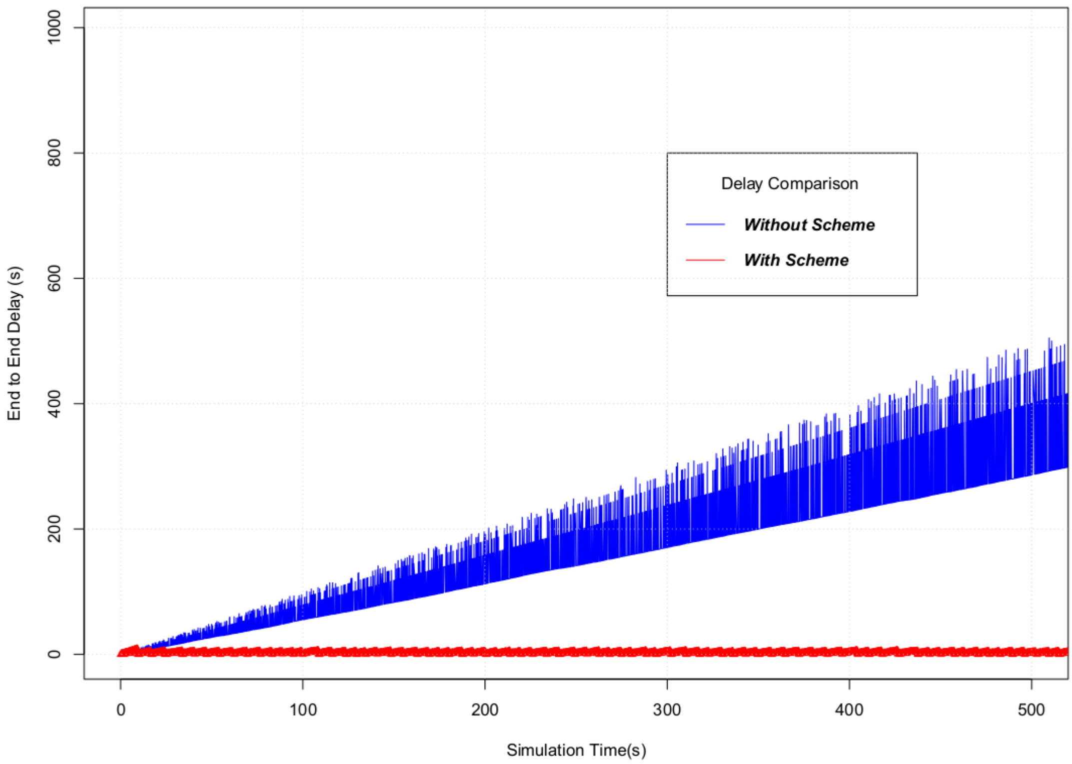

Figure 6.15 End to End Delay Improvement

6.5 Conclusion of Chapter

This chapter investigated end to end latency experienced by high priority packets. Also the Buffer size level at the relay node and also the source node were observed .The scheme was discussed on the possible ways of improving latency and application of the scheme to IoT networks.The scheme relies on neighbouring nodes having extra memory or storage where packets from a relay node can be sent during times of congestion or near congestion.Simulation shows some promise in implementation however a lot of factors still to be considered such as how surrounding nodes exchange buffer length infomation.

Chapter 7

Conclusion and Future work

This work studied the implementation of a virtual buffer extention for packet delay reduction considering different network topologies.The network considered is a multihop network with intermediate basestation as relay node.All base station had mesh clients attached to them at the same time relaying traffic for other basestation nodes.In resource constrained environment,in case of finite and small size buffer,this can cause packet drops.To avoid this packets to be lost ,buffer extension is studied.The main purpose of having virtual buffer,the whole premise of this investigation is based on the probability of blocking in network nodes. Dependability is achieved by reducing this plocking probability.The wireless network environment has low power devices which interconnect and these are enablers for IoT.With the anticipated coming of 5G,it is anticipated a lot of devices will be connected to one form of a network or the other.

7.1 Future work

Most of these devices such as wireless sensor networks have constrained resources,amongst them being memory constraint.With the introduction of a scheme to distribute storage around the surrounding nodes,priority can be given to packets of a higher class.Instead of losing packets.The premise of this thesis is on various subjects which are interdiscplinary within the communication space.There still many issues related to studying of Virtual buffer extenstion in networks.Some of them include protocol design and how signalling or messaging flows.Other areas of future research is in the ability of the network nodes to adapt to any role.The heiriechical structure is such that we have layer 1 nodes and layer 2 nodes.layer one nodes being at base station level,while layer 2 nodes being at client level.The ability of the roles to be interchangeable would be an interesting subject to study. An investigation of the flow of different traffic types modelling real world traffic in the network is another future area of study. The other area which is to be investigated is the effect of changing the parameters of the reed solomon code,in this thesis a fixed size was used for all encoding i.e RS (n,k) was RS (255,223).Therefore the message or packet size needs to be varied for various sizes of n and k which are the encoded packet size and the message size.

Published Papers

Reviewed Journal Papers

(1) Zilole Simate, Ryuji Kohno, “Investigating Buffer Extension in Multi-Hop Networks for Packet Delay Improvement,” International Journal of Computer Science and Telecommunications , (IJCST) ISSN 2047-3338 (Published).

(2) Zilole Simate, Ryuji Kohno, “A combined Network Coding and Routing strategy in Low power Wide area Networks,” IEEE Systems Journal, (to be submitted).

(3) Zilole Simate, Ryuji Kohno, “ Routing strategy considering finite buffer in sensor networks, ” Sensors Journal” Sensors Journal, (to be submitted).

International Conference Papers

(1) Zilole Simate, Ryuji Kohno “Investigating an Adaptive Network Coding Control Protocol for Multihop-Multipath Networks for Network Reconfigurability,” International Conference on Computer and Communication System (ICCCS2018), Nagoya, Japan, April 2018.

(2) Zilole Simate,Chika Sugimoto, Ryuji Kohno “Protocol for Improved Network Coding Opportunity Discovery for Inter-connected WBAN Multihop Relay Medical Networks,” International Symposium on Medical Information Communication Technology (ISMICT 2019),Oslo,Norway, May,2019.

(3) Zilole Simate “Investigating the use of Interactive Voice response(IVR) in Medical Monitoring Adherence” International Symposium on Medical Information Communication Technology (ISMICT 2014),Firenze,Italy, March,2014.

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