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Advanced Journal in Wireless and Mobile Communication

Aims and Scope

Advanced Journal in Wireless and Mobile communication welcomes the original research papers, review papers, experimental investigations, surveys and notes in all areas relating to software engineering and its applications. The following list of sample-topics is by no means to be understood as restricting contributions to the topics mentioned:

•LTE
•4G
•5G
•Ultra wide band communications
•Nnovel mobile applications
•Mobile communications and networking
•Energy-efficient communication networks
•Molecular communications

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Contents

Data Give-and-Take in Wireless Networks with Network Coding Mechanism

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ABSTRACT

In this paper we have analysed the different types of Information exchange in wireless network that includes Unicast, Broadcast & Multicast &elaborate about the role of Network coding in such information exchanges. Recently it has been shown that network coding improves network reliability by reducing the number of packet retransmission. Network coding is a developing method which is currently pragmatic to wireless networks to improve network throughput and other performance. Contemplate cooperative wireless network in which there are several sources & several relays. The unreliable wireless channels, the quality of network links between nodes can vary; which result in the failure of intermediate nodes which results in to the linear combination of incoming massage in network coding scheme. At base station we have proposed the recovery performance of sources massages.

Keywords: Multicasting, Network coding, Network throughput

1. INTRODUCTION

Information exchange can be done in three different ways these different scenario are Unicast, Multicast & Broadcast. Unicast is the term used to describe communication where a piece of information is sent from one point to another point. In this case there is just one sender, and one receiver. Broadcast is the term used to describe communication where a piece of information is sent from one point to all other points. In network case there is just one sender, but the information is sent to all connected receivers. In applications like distributing weather reports, stock market updates or live radio programmes, broadcast is the best way to spread information. Broadcast is mostly used in local sub-networks. Multicast is the term used to describe communication where a piece of information is sent from one or more points to a set of other points. In this case there is may be one or more senders, and the information is distributed to a set of receivers. Below diagram shows overview. Multicast is a network addressing method for the delivery of information to a group of destinations simultaneously using the most efficient strategy to deliver the messages over each link of the network only once, creating copies only when the links to the multiple destinations split (typically network switches and routers). Multicast is often used for streaming media and Internet television applications.

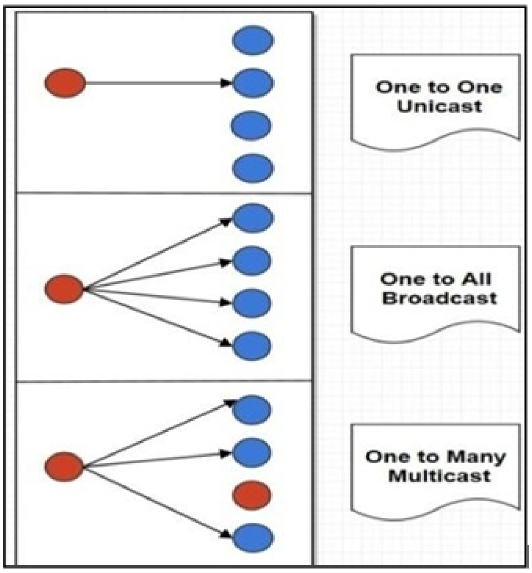

Fig1.Overview of Unicast, Broadcast Multicast

Network Coding is a field that was first introduced in 2000 [5] as a method to utilize the maximum capacity of a network and maximize the flow of information in that network. It suggested coding at packet level in wired P2P networks. The idea sprouts from research done in[4] on satellite communications using a source coding system which consists of multiple sources, encoders, and decoders.

Network coding is a networking technique in which transmitted data is encoded and decoded to increase network throughput, reduce delays and make the network more robust. Network coding is perceived to be useful in wireless mesh networks, messaging networks, storage networks, multicast streaming networks, file-sharing peer-to-peer networks and other networks where the same data needs to be transmitted to a number of destination nodes. The regular topology change that occurs in peer-to-peer networks poses a challenge to the network coding technique because it complicates network synchronization. In addition, the peers may need a large amount of processing time while trying to decode data. Network coding promises significant benefits in network performance, especially in lossy networks and in multicast and multipath scenarios.

2. DETAILS EXPERIMENTAL

2.1. Multicast Topology

As shown in Fig2 When a sender has the first multicast packet to send, it launches an initiating process which includes assigning FEC to packets flow, computing sending interval according to various application mode, and generating a new FEC table item, etc. Then, it generates and sends a Label

-Request message through flooding. During the setup process, not only a multicasting tree but also a mesh is generated. These two kinds of forwarding topology can work alternatively to match the varying network environment. This is made possible by down-stream nodes sending Label-Mapping message to up-stream nodes with two different modes: unicasting via inverse path and flooding. Nodes which have received the Label-Mapping message can potentially become forwarding nodes of the multicasting mesh. Thus, the multicast topology is determined by destination nodes according to their judgments to the network conditions, and source nodes have no need to maintain the multicast topology.

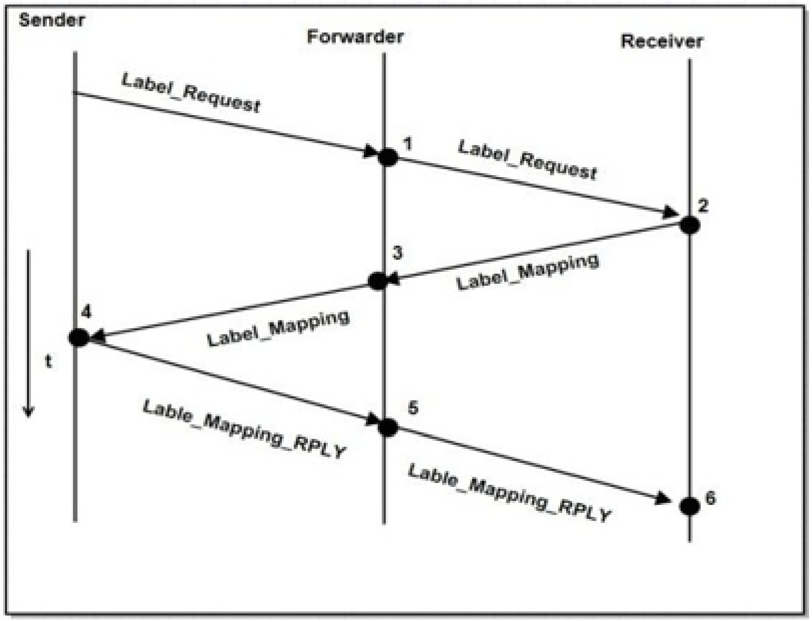

Fig2. Multicast Topology

2.2 Algorithm

Within the scope of the network coding literature, a number of papers have proposed algorithms that employ network coding over a dynamically changing wireless environment & evaluated their performance through simulation results closest to our particular broadcasting problem is [6] which shows that from the viewpoint of packet delivery ratio & overhead, NC compares very favourably to flooding. Minimum cost multicasting using network coding was examined in [1] for mobile networks and in [2] for fixed networks. Our work differs in that, rather than solving the routing problem we focus on assessing the benefits network coding may offer.

3. RESULTS AND DISCUSSION

3.1. Performance Evaluation for Multicast member Node s

The number of duplicate data packet as a function of the number of multicast member nodes with node mobility of 20km/h. As we can see in Fig 3, CRMP generates significantly lower number of data packet transmissions than ODMRP. As the number of multicast member nodes increases, the gap in the number of duplicate data packet of our proposed CRMP and ODMRP is larger. CRMP has moderately lower number of data transmissions because it can reduce the number of forwarding node by using optimal route refresh interval calculated from the information of node mobility in the network.

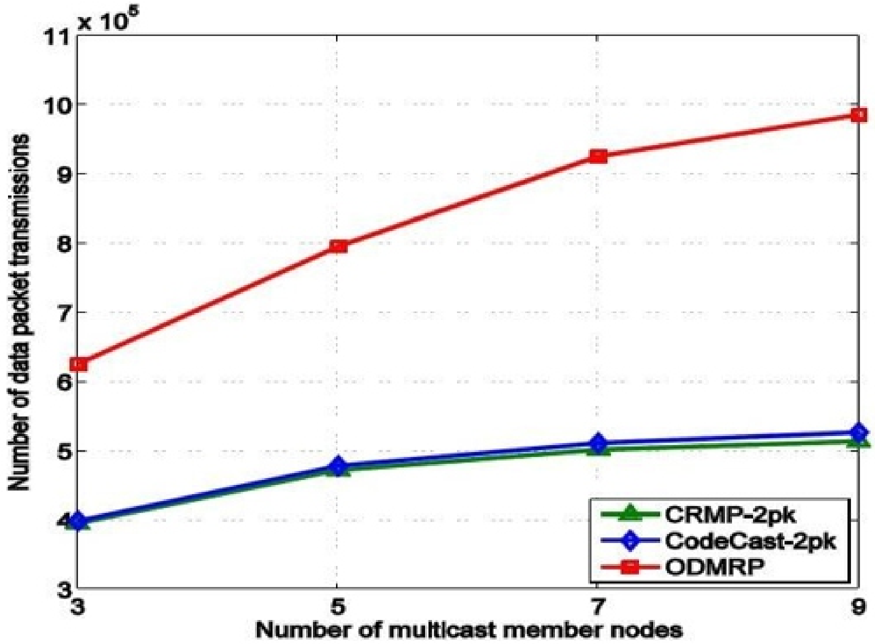

Fig3.The Number of Duplicated Data Packet as a Function of the Number of Multicast Member Nodes

3.2 Network Coding Max. Flow

Network coding is a recently proposed mechanism based on a simple idea first stated by Ahlswede et al. [4]. In fact, it was shown [5] that if we regard multicast throughput as information flow, then max-flow of point-to-multipoint communication is:

Where F (t) is a max-flow from source s to t, and M is the set of nodes which are multicast receivers. Furthermore, multicast max-flow cannot be achieved when using IP multicast,but it is achievable when using net work coding [3].

$$F_{MCAST} = \min_{t \in M} F(t) \qquad \text{I}$$

CONCLUSIONS

In this paper, we have studied the Network data transfer mode & basic working. We also showed that Network coding mechanism is effective, method for multicasting for throughput optimization. Also we have proposed a robust Network Coding based Multicast Routing. Further research should focus on whether network coding benefits can be used by practical systems in realistic settings.

ACKNOWLEDGMENTS

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Performance Analysis of Heirarchical Routing Protocol in Wireless Sensor Network

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A B S T R A C T

The Wireless Sensor Network (WSN) is a wireless network which consists of number of small nodes with sensing, computing and wireless communication capabilities. Usefulness of WSN can be realized from the fact that it is widely used these days to monitor activities and report events, such as fire, overheating, environmental conditions etc. in a certain area. Many protocols have been designed due to recent advancement in wireless sensor network. These protocols are used to lower energy consumption. Thus development of an energy efficient routing protocol has interested researchers. There are many hierarchical routing protocols among them author simulated leach protocol via ns2 and analyze the performance in terms of rounds, throughput and data sent to the base station.

Keywords: LEACH, Wireless Sensor Networks, NS2, Sensor Node Energy

INTRODUCTION

A wireless sensor network (WSN) may be defined as a collection of sensor nodes that usually derive their energy from attached batteries **Bilal Abu Bakr and Leszek T. Lilien, 2014 .**The sensor node is smart which can communicate in between or to the externally located base station. Wireless sensor networks are used for several applications such as traffic monitoring, surveillance, acoustic and seismic detection, environmental monitoring, etc **Salim el Khediri et al., 2014**. Wireless sensor network provides a large range of potential application to industry, Science, civil infrastructure and security.

WSN lifetime is the key characteristics for the evaluation of sensor networks. In WSN the main concern is how we can save energy of the sensor nodes. For this there are so many algorithms and routing techniques are developed to save the sensor node energy as much as possible to increase the lifetime of the networks. It is impossible that all the nodes directly communicate to the base station because of stringent constraints **Geetha.V. et al.,2012.** and nature of radio communication. As we discussed before, in WSN energy is the main constraint if communication is hop by hop then there is a lot of energy wasted which drastically reduce the life time of the network. To overcome this problem clustering introduced which creates a balance among the key factors of the WSN node operation simultaneously. Clustering is defined as the process of selecting one node from the group of node to act as a servicing node for the neighbor nodes. Hierarchical routing is required when the size of the WSN increases for example if the application has thousands of nodes then it is preferable to have a three level or four level hierarchy so that the lifetime of the network increases.

The challenges that wireless sensor network have

- Energy Efficiency
- Responsiveness
- Robustness
- Self-Configuration and Adaptation

RELATED WORK

Routing is a process of selecting a path through which data can be transmitted in the network from source to destination. Protocols such as LEACH, HEED, PEGASIS, TEEN and APTEEN are the hierarchical routing protocols used to route the data from node to base station.

Sensor nodes organize themselves into clusters and each cluster has at least one cluster head which act as a leader of the cluster. In this Network where low energy nodes sends the collected information to the cluster head after collecting the information from the nodes the cluster head remove the redundant information and then send data to the base station which increases the life time of the network .the process of clustering in routing provides an efficient technique to increase the life time of the network by rotating the role of the cluster head.

LEACH

Low energy adaptive routing protocol **Wendi Rabiner Heinzelman et al., 2000.** introduced by W. Heinzelman is a hierarchical routing protocol in which nodes transit the collected data to the cluster head and then cluster head compresses this data by eliminating redundancy and then send this data to the base station. Leach protocol works in rounds. In each round Leach protocol chooses the cluster head. The network model of leach is shown in figure 1 as shown in network model that all the nodes directly communicate with the cluster head and the cluster head is then directly communicated with the base station. Leach uses hop to hop communication means all nodes within the cluster should directly communicate with the cluster head. It uses an algorithm for selecting the cluster head.

Two phases are in LEACH protocol which is (i) the cluster formation and (ii) data receiving and transmission phase and round as defined the time slot gap between two phases Raju Dutta et al., 2013. In the cluster formation phase the process of cluster header selection is that the sensor node generates a random number between 0 and 1, If it is less then threshold T (n), it will selected as a cluster header, and report to the other nodes.

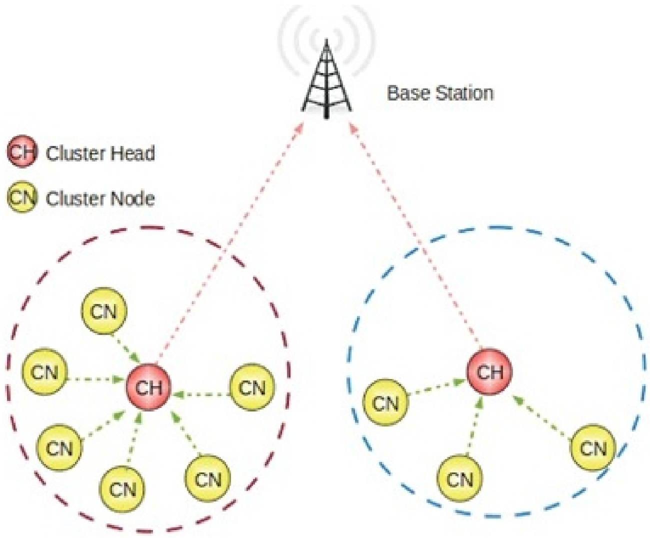

Figure 1: Network Model of LEACH D.Suresh and K.Selvakumar, 2014

The T (n) equation is as following Jianguo Shan et al., 2013

$$T_n = \begin{cases} 0 \; if \; n \notin G \\ \dfrac{P}{1-P(r \; mod \; (\frac{1}{P}))} & \text{for all n belongs to G} \end{cases}$$

r is the current round, p is the probability to become the cluster header, and G is the number of nodes that were not cluster headers in (r-1) round. N is the total number of nodes, k is the expected number of cluster headers, then p = k/N .This algorithm can make each sensor node become the cluster header each once, if one node is cluster header in the round, it will not be in the next round.

The Work Process of Leach Protocol

Leach works in rounds and the round begins with the setup phase, where clusters have been organized followed by steady state phase where data is send to the base station. To reduce the overhead the steady state phase is longer compared to setup phase.

Cluster Establishment Phase: after selecting the cluster head, Mac protocol is used to broadcast the ADV news to all nodes. ADV news includes node ID and packet header used as a identifying news type

LiTian et al., 2012.the node receives a signal and according to the signal strength it decides to join the cluster for this it uses MAC protocol to send a request (JOIN-REQ) to the corresponding cluster header. The JOIN-REQ includes node ID, cluster header ID and packet header **LiTian et al., 2012.** After cluster has been organized it creates a TDMA schedule based on the number of nodes belongs to the cluster and broadcast this TDMA schedule to all the nodes within the cluster t when to transmit the data according to their time slot. The TDMA mechanism efficiently transmits the data within the cluster and avoids the conflict during communication in the cluster.

Data Transmission Phase: After cluster has been organized and TDMA schedule is set, data transmission can start. This phase lasts a long time in the wireless sensor network. In this phase assuming all nodes have data to send but the nodes send their data according to the given time slot after sending the data the nodes will go into sleep and wait for his turn but receiver of the cluster header keep open so that it can receive the data which is send by its member at any time. Communication within the cluster would inescapably affect the others. To avoid this CDMA is used. When a node becomes cluster head it will choose a code and then send a code with the TDMA schedule. So the communication within the cluster cannot affect. After collecting all data from the member nodes the cluster head fuses all data and compresses this data by eliminating correlated data and send it to the base station. The work process of Leach protocol is shown in figure above.

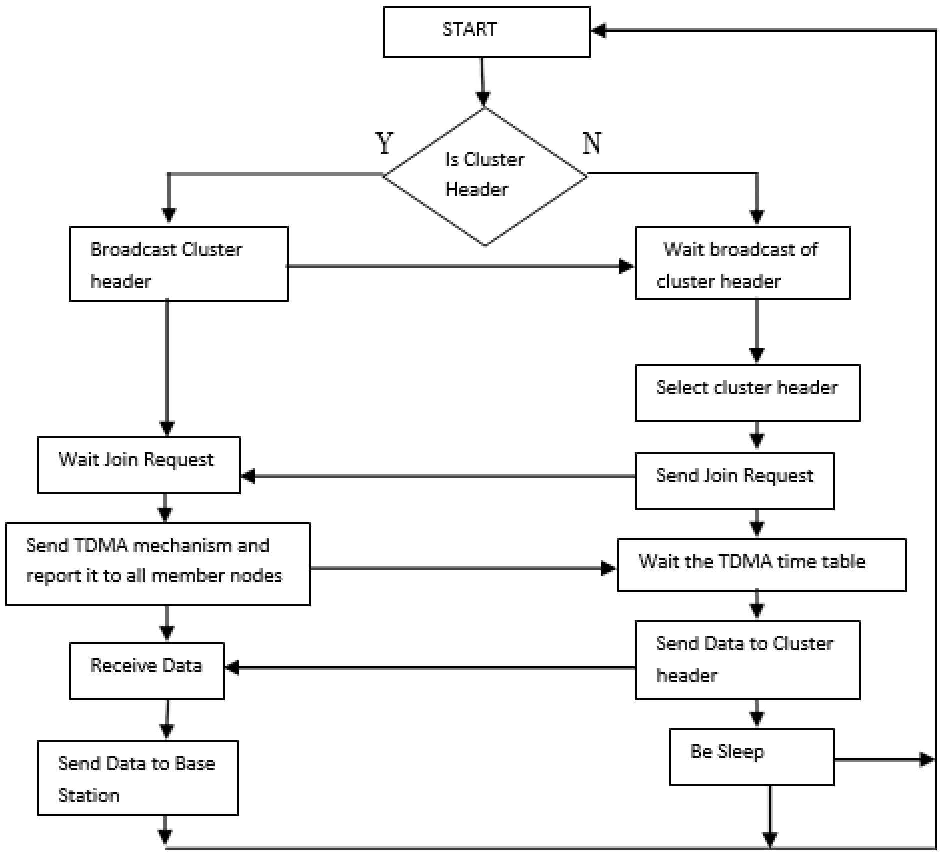

Figure 2: The work process of LEACH protocol LiTian et al.,2012

The Simulation and Analysis of Leach Protocol

Results of the simulation are presented in Table 2, Table 3, Table 4 and Table 5. Table 2 shows the data sent to the base station with respect to time at 2 Joule energy which lasts up to 2580s. Table 3 shows the data sent to the base station for 4 Joule energy which lasts up to 2720s. Table 4 shows the throughput at different no. of nodes and at different energy. Table 5 shows the life of the network in terms of rounds on different no. of nodes and at different initial energy. Figure 3 and Figure 4 shows the data sent to the base station with respect to time at 2 Joule and 4 Joule energy respectively. Figure 5 shows the life of the network in terms of rounds with respect to nodes at 2 Joule and 4 Joule energy.

Table 1: Simulation parameters

Sl. No	Item Description Specification	Item Description Values
1	Simulation Area(m*m)	100*100
2	No of Nodes	5,10,15,20
3	Channel	Wireless Channel
4	Propagation	Two Ray Ground
5	Netif	Phy/wireless phy
6	Mac	Mac/802_11
7	Protocol	Leach
8	Ifq	Queue/Drop Tail/PriQueue
9	Ll	LL
10	Antenna	Omni Antenna

Table 2: Time - Data send to base station (2 Joule)

Time(s)	5 nodes	10 nodes	15 nodes	20 nodes
500	11916000	12124000	13320000	15660000
1000	15428000	15868000	17836000	24892000
1500	16412000	16960000	19260000	30264000
2000	16632000	17220000	19644000	32560000
2140	16660000	17248000	19672000	32828000
2220		17264000	19688000	32936000
2280			19700000	32996000
2580				33148000

Table 3: Time - Data send to base station (4 Joule)

Time(s)	5 nodes	10 nodes	15 nodes	20 nodes
500	23624000	24024000	26472000	30876000
1000	30920000	31796000	35884000	49580000
1500	33044000	34156000	38900000	60512000
2000	33608000	34812000	39780000	65236000
2440	33720000	34960000	39960000	66452000
2540		34980000	39980000	66556000
2560			39984000	66572000
2720				66632000

Table 4: Nodes- Throughput

Nodes	2 Joule	4 Joule
5	150898.16	152203.16
10	629400.25	639821.84
15	1443217.13	1446801.58
20	1840826.65	1843644.8

Table 5: Nodes - Rounds

Nodes	2 Joule	4 Joule
5	107	122
10	111	127
15	114	128
20	125	136

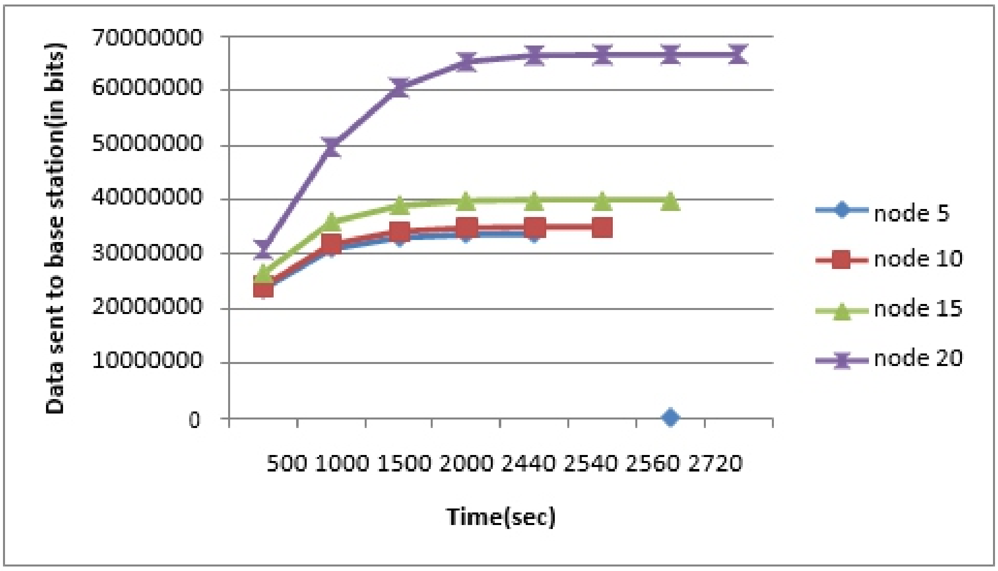

Figure 3: Data sent to base station (bits) vs. time (sec)

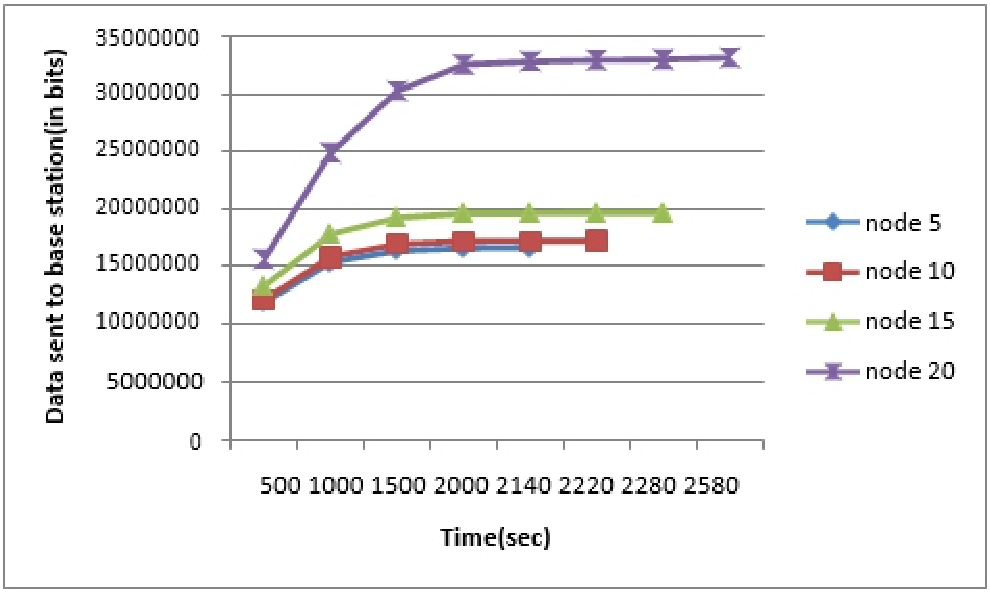

Figure 4: Data sent to base station (bits) vs. time (sec)

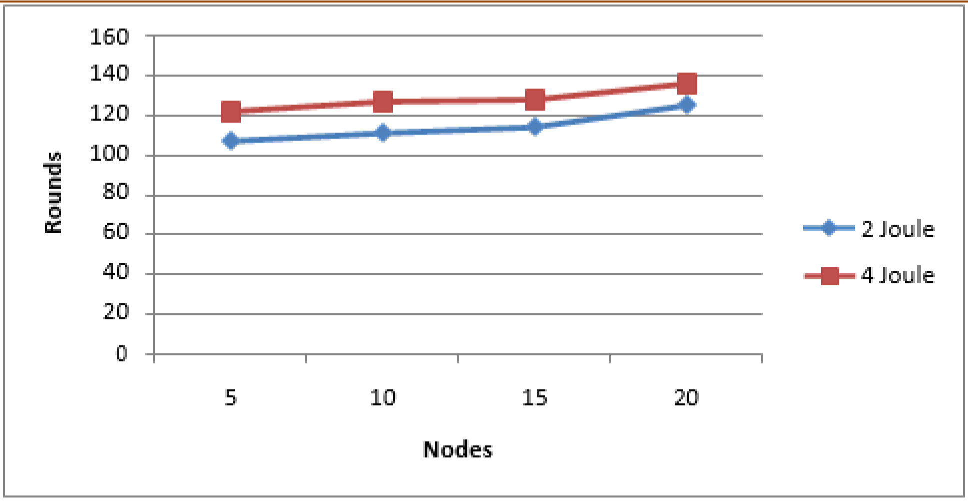

Figure 5: Nodes vs. Rounds

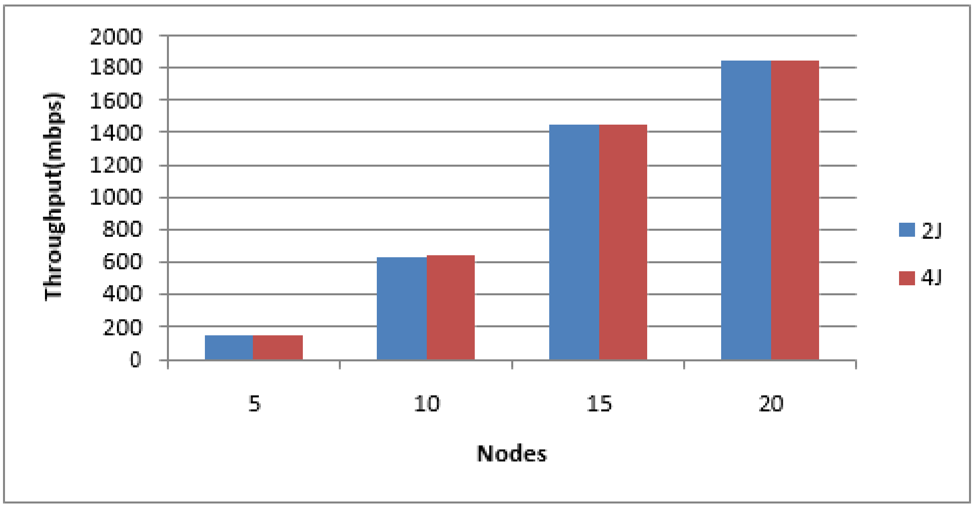

Figure 5: Nodes vs. Throughput (Mbps)

CONCLUSION

From the simulation results we conclude that on increasing the number of nodes, the lifetime of the network in terms of rounds, the throughput of the network and the data sent to the base station increases. Increase in the initial energy of the nodes from 2 Joule to 4 Joule results in increase of lifetime of the network in terms of rounds and data sent to the base station also increases with this increase in initial energy. On the other hand there is less increase in the value of network throughput.

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Performance Evaluation of Mobile WiMAX using Sliding Window MMSE Equalization

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A B S T R A C T

WiMAX is introduced by the Institute of Electrical and Electronic Engineers (IEEE) which is standard nominated 802.16d-2004 (used in fixed wireless applications) and 802.16e-2005 (mobile wireless) to provide a worldwide interoperability for microwave access.). IEEE 802.16e-2005 has been urbanized for mobile wireless communication which is based on orthogonal frequency division multiplexing (OFDM) technology and this enables going towards the 4G mobile in the future.

In a usual OFDM broadband wireless communication system, a guard interval using cyclic prefix is included to avoid the inter symbol interference (ISI) and the ICI. This guard interval is necessary to be at least equal to, or longer than the maximum channel delay spread. This method is very simple, but it reduces the transmission efficiency.

To compensate affect of ICI by applying equalizers at receiving side is an active research area. Zero forcing (ZF) and Minimum mean square error (MMSE) are mainly two equalization algorithm. Unfortunately, the MMSE method requires the inversion of an N × N ICI matrix, where N is the number of subcarriers. When N is large, the computational complexity can become prohibitively more. Utilizing the fact that ICI energy is clustered in adjacent subcarriers, MMSE equalization is made localized.

The aim of this paper is the performance evaluation of 802.16e OFDM-PHY system using sliding window MMSE channel equalizers at the receiver side for different doppler frequency. The Bit Error Rate (BER) of the wireless communication channel using Recursive sliding window Minimum Mean Square Error (MMSE) equalizer is analyzed. The simulation includes BER versus Energy to noise ratio (Eb/N0) at various doppler frequencies for performance predictions.

Keywords: OFDM, ISI, ICI, MMSE Equalizer, ZF, BER, Mobile WiMAX.

INTRODUCTION

Worldwide Interoperability for Microwave Access (WiMAX) is based on wireless metropolitan area networking (WMAN) standards developed by the IEEE 802.16 group and adopted by both IEEE and the ETSI HIPERMAN group [1]. It provides very high data throughput over long distance in a point-to multipoint and line of sight (LOS) or non-line of sight (NLOS) environments. WiMAX can provide seamless wireless services up to 20 or 30 miles away from the base station [2].

The IEEE 802.16e-2005 forms the basis for the WiMAX solution for mobile applications and is often referred to as mobile WiMAX [3].

OFDM technique is widely adopted in those systems due to its robustness against Multipath fading and simpler equalization scheme. In most of applications, for retaining the orthogonality of subcarriers and overcome ISI, a cyclic prefix (CP) is inserted instead of simply inserting guard interval. If the maximum delay of the Multipath channel does not exceed the CP length, the OFDM system would be ISI free by removing the guarding interval [4]. For WiMAX systems, its delay spread is typically over several microseconds which are longer than the guarding interval.

Therefore, it is very challenging to maintain the system Bit Error Rate (BER) performance for NLOS channels at high data rate transmission also for mobile WiMAX. Doppler Effect degrades system performance. Both, the equalizer or channel estimator can be applied to compensate for the attenuation and phase shift introduced by the channel [5]. Equalization and channel estimation is simple for OFDM systems but it needs careful consideration due to their implementation limitations to accomplish the trade-off between complexity and accuracy.

INTER CARRIER INTERFERENCE (ICI):-

Radio channel are arbitrary, fast varying and inaccuracy prone. In a wireless system the variation/oscillation in the received signal is called fading. The aim of the wireless system design is to overcome different types fading and offer consistent and competent transmission. Generally there are two types of fading [6]:

- Large scale fading: It is the instability in the average signal strength over a large distance and is caused by earthly change. This occurs when a mobile travel from a lake tohilly area or from an open area to a high buildings area. Large scale fading can be mitigated by controlling the transmit power.
- Small scale fading: Occurs as a effect of the fluctuations in the received signal strength over a little distance and is caused by multipath and Doppler's shift. Doppler shift refers to the alteration on frequency of the signal because of comparative motion between the transmitter and the receiver.

The Doppler shift is the variation in frequency and wavelength of a wave for an observer moving comparatively to the source of the waves. In multipath fading channel, different Doppler shift on every of the multipath components guide to random frequency modulation as long as there is relative motion between the base station and the mobile, Doppler shift is given by [6].

$$f_d = (v/\lambda)\cos\theta$$

where: Above Equation refers to the relation between the Doppler shift and the mobile velocity as well as the spatial angle between the direction of mobile motion and the wave arrival.

The Doppler shift introduces a different type of interference in OFDM i.e. ICI [7]. OFDM separated the spectrum into narrowband subcarriers and they are closely spaced simply because they are orthogonal. One of the requirements for orthogonality is to preserve the subcarrier spacing exactly the reciprocal of the symbol period [7]. The frequency shifts thus varying the subcarrier spacing which results in the loss of orthogonality. This loss of orthogonality creates interference between the signals which is called as ICI [7]. Since the subcarriers in OFDM are usually very narrow hence the OFDM system becomes very sensitive to ICI. ICI destroys the orthogonality of the OFDM system which is overcome by the use of cyclic prefix method.

Researchers have proposed various methods to combat the ICI in OFDM systems:

- **Frequency Domain Equalization**
- **Time Domain Windowing**
- **Pulse Shaping.**
- **ICI Self Cancellation**

Equalizer at receiver side to improve the performance in time selective channel is allowed in Mobile WiMAX [8]. Equalizer in wireless communication is adaptive traversal linear filterwhich is designed as channel compensator. The value of weight of equalizer is adjusted according to equalization algorithm. ZF and MMSE equalization algorithm are two most popular algorithms [9].

SYSTEM MODEL

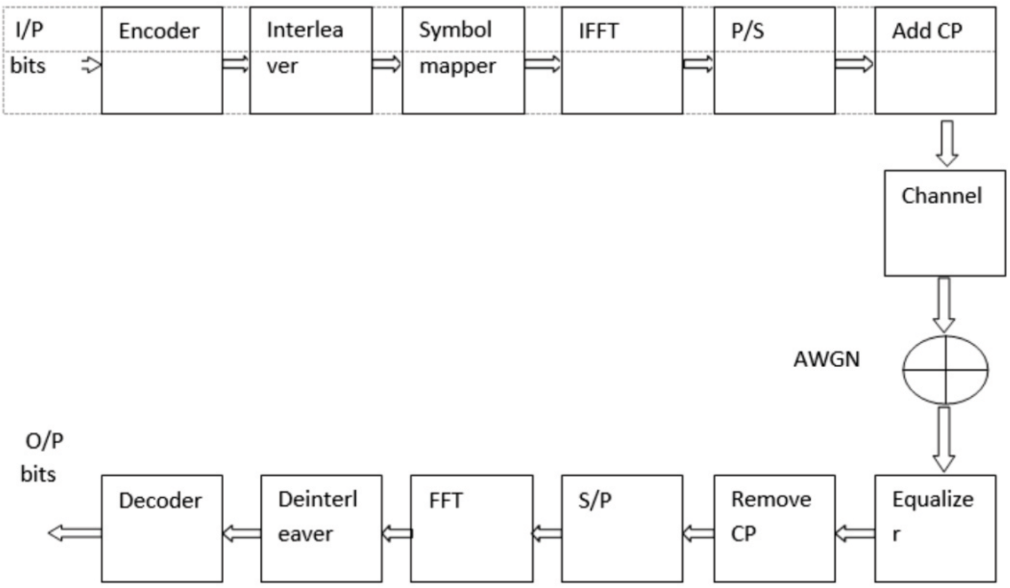

Fig.1.OFDM based physical layer system of mobile WiMAX

The OFDM system model is

$$Y = HX + w \qquad (1)$$

Where, $Y = [Y_0, Y_1, \ldots, Y_{N-1}]^T$ are the received OFDM subcarriers.

$X = [X_0, X_1, \ldots, X_{N-1}]^T$ are the transmitted OFDM subcarriers.

$w = [w_0, w_1, \ldots, w_{N-1}]^T$ are the additive white Gaussian noise.

$$\text{And } H = \begin{bmatrix} H_{0,0} & H_{0,1} & \ldots H_{0,N-1} \\ H_{1,0} & H_{1,1} & \ldots H_{1,N-1} \\ H_{N-1,0} & H_{N-1,1} & \ldots H_{N-1,N-1} \end{bmatrix} \qquad (2)$$

As equalizers are compensator for channel distortion so the equalizer output is given by:

$$X = GY \ (3)$$

Where, G is the N-by-N equalizer matrix which minimizes the cost function $E\left\{|X - \hat{X}|^2\right\}$ and the solution is,

$$G = [R_{xy} R_y]^{-1} \qquad (4)$$

The resulting MMSE is then given by :

$$MMSE = Tr(R_x - R_{xy} R_y R_{yx})^{-1} \qquad (5)$$

Where, $R_{xy} = \sigma^2_x H^H$, (6)

$R_y = \sigma^2 xHHH + \sigma 2wIN$, (7)

Where R denotes covariance matrix, the superscript H denotes complex conjugate transpose and IN is the N-by-N identity matrix

Then (3) can be rewritten as

$$G = H^H (HH^H + I_N) \qquad (8)$$

Likewise (4) becomes $MMSE = \sigma^2_x Tr(1 - GH)$ (9)

SLIDING WINDOW MMSE EQUALIZER:

Using the fact that ICI energy is concentrated in adjacent sub channels, the complexity of the frequency domain equalization can be significantly reduced without much performance degradation. The MMSE equalizer is too complex to be implemented, especially when N is large. Using the fact that the ICI power is localized to the neighborhood of a desired sub channel, only a few neighborhood sub channels can be used for equalization without much performance penalty [10]. If the ICI channel memory has the correlation length among each data subcarriers is at most 2q. Recursive Sliding window MMSE equalizer employs a much smaller window of size $(2q+1)\times(2q+1)$

More specifically, the channel matrix Hm ε C(2q+1)×(2q+1) in sliding window MMSE equalizer becomes

$$\mathbf{H}_m = \begin{bmatrix} h_{m-q,m-q} & \dots h_{m-q,m+q} \\ h_{m,m-q} & \dots h_{m,m+q} \\ h_{m+q,m-q} & \dots h_{m+q,m+q} \end{bmatrix} \qquad (10)$$

The problem is to find the equalizer coefficient vector $g_m = [g_m, 0, \dots, g_m, q-1]$ From (7), the MMSE solution is

$$g_m = R_{XmYm} R_{Ym}^{-1} \qquad (11)$$

And, by considering the same assumption as in the previous section, the solution for MMSE equalization is given by:

$$R_{XmYm} = E\{x_m y_m^H\} \qquad (12)$$

$$= \sigma x2hmH$$

Where hm is the mth column of the matrix H_m

Also $$R_{Ym} = E\{y_m y_m^H\}$$

$$= \sigma_x^2 H_m H_m^h + \sigma_w^2 Iq. \ (13)$$

After inserting (12) and (13) into (11), the q-tap equalizer vector gm becomes

$$g_m = h_m^H (H_m H_m^h \frac{\sigma_w^2}{\sigma_x^2} Iq)^{-1} \qquad (14)$$

Similarly, $$MMSE = \sigma_x{}^2 \sum_{m=0}^{N-1} (1 - g_m h_m) \qquad (15)$$

By choosing an appropriate number q, the complexity of the equalizer can be reduced significantly.

RESULTS AND ANALYSIS

Simulation results of the proposed equalizers with application to IEEE 802.16e mobile WiMAX standard are evaluated. The input data is encoded with 64- state rate-1/2 binary convolutional code (BCC) with polynomials in octal notation $(133,171)_8$, 16-QAM modulation is used with 256 subcarriers. The channel considered for simulation is COST-TU with 12 numbers of taps. The normalized Doppler frequency f_dT_s= 6.82% and 13.64% is applied one by one. At receiver side MMSE equalizer is applied with 5 numbers of taps and 11*11 sized windows is being used for channel estimation. For analyzing performance of proposed algorithm between BER and E_b/N_0 at various f_dT_s

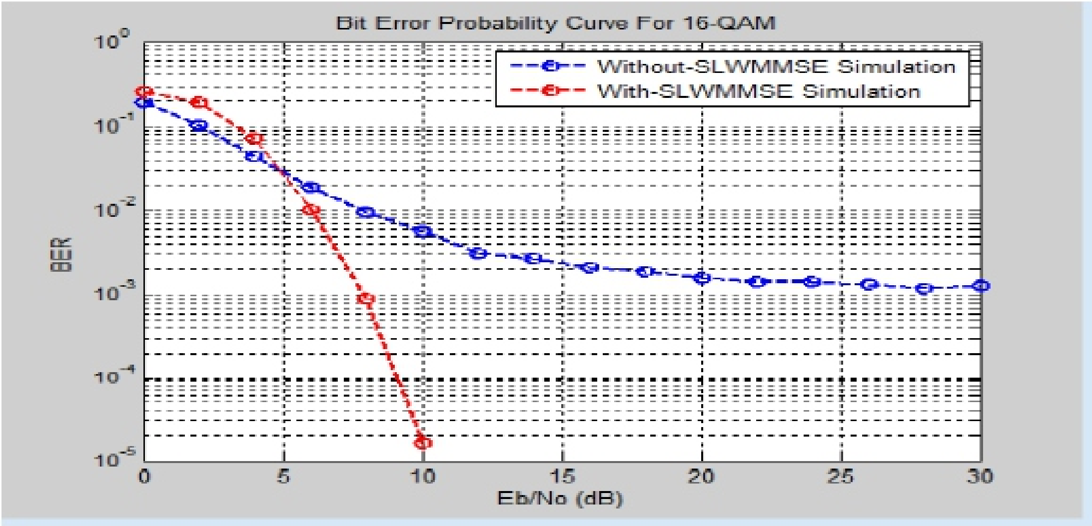

Fig.2. BER of OFDM based physical layer mobile WiMAX with and without sliding window MMSE equalization for f_dT_s= 6.82%.

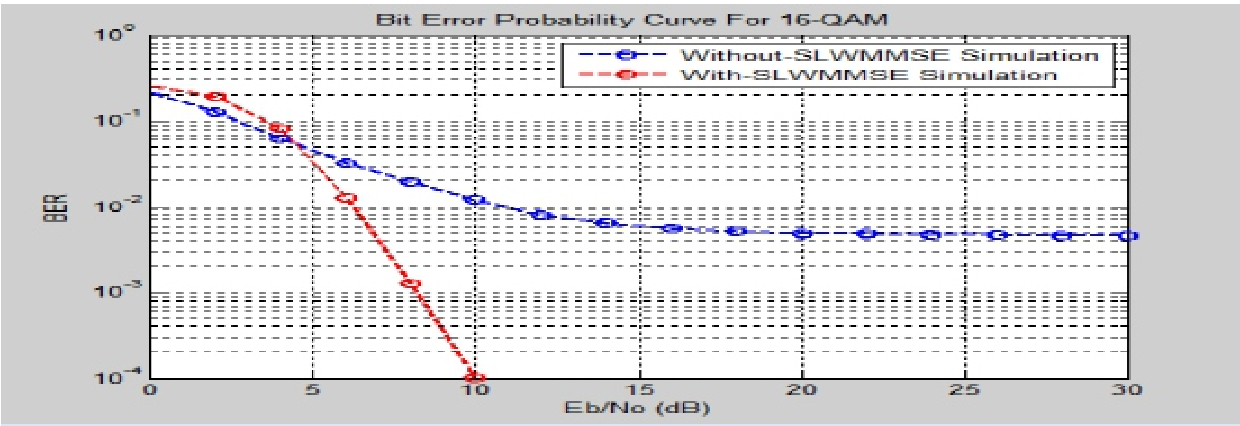

Fig.3. BER of OFDM based physical layer mobile WiMAX with and without sliding window MMSE equalization for fdTs= 13.64%.

By analyzing the result this can be concluded that SLW MMSE equalizer offers same BER at low values of E_b/N_0 for different f_dT_s . At E_b/N_0=10dB the proposed SLWMMSE equalizer gives BER $1.63*10^{-5}$, $1.02*10^{-4}$ at fdTs =6.82%, 13.64% respectively. This is because higher value of f_dT_s introduces more ICI in the system.

Due to energy concentration property of ICI the proposed SLW MMSE used 5 numbers of taps without performance degradation.

CONCLUSION

As demand for high speed communications under different mobile scenarios rises, the ICI problem of OFDM systems become an important issue. Iterative equalization and decoding of the wireless mobile coded OFDM system is considered. Complexity reduction compared to the MMSE equalizer is achieved due to the energy concentration property of ICI. Computational efficient recursive sliding window MMSE based equalizer are derived which provides excellent performance and complexity trade-off. The performance of the equalizer has been evaluated over the COST-TU channel model with application to WiMAX. The results suggest that iterative processing at the receiver end allows full exploitation of both temporal and frequency diversity available in a spectrally efficient system.

This is concluded that OFDM based physical layer of mobile WiMAX system applying recursive sliding window MMSE equalization at receiver side having 5 number of tap with 64 state BCC encoding scheme, QAM-16 modulation with 256 subcarrier, with COST –TU channel having 12 number of taps and applying fdTs =6.82%, 13.64% one by one gives reduced BER as compared to other techniques. As the normalized Doppler frequency increases, the performance of proposed algorithm degrades due to higher ICI in the system.

FUTURE WORK

The proposed coded-OFDM signal reduces the BER with recursive sliding window MMSE equalization at receiver side. Thus future work is with MIMO system.

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A Distributed Call Admission Control in Cellular Mobile Networks using Intelligent Techniques

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ABSTRACT

Quality of Service (QoS) provisioning in wireless mobile networks is quite a challenging task due to the scarcity of wireless resources. With the future wireless/mobile networks moving towards micro/pico cellular architecture, it would become all the more challenging owing to the frequent handoffs. Call admission control (CAC) is one of the key elements in ensuring QoS in mobile networks. The present paper focuses on distributed call admission control (DCAC) scheme, in which the call admission decision uses the information not only from the local cell but also from the neighboring base stations and the information is exchanged periodically. It has been shown by various researchers that a call admission scheme based on distributed approach can guarantee QoS requirements. In this regard, the paper presents fuzzy logic based distributed call admission control (FDCAC) scheme. Further, the scheme takes into consideration of diverse traffic types and also considers the priority of calls in order to provide higher connectivity to high priority users. In this paper, two variants of FDCAC distributed approach have been considered and their performance has been compared with that of the simple guard channel scheme in terms of various QoS parameters such as new call blocking probability and call dropping probability. Based on the results, the bandwidth efficiency of the proposed scheme has been verified and compared with the other models including guard channel scheme.

Keywords: Quality of service, distributed call admission control, call admission control, Fuzzy logic, cellular mobile networks.

1. INTRODUCTION

Recent years have witnessed explosive growth in the field of wireless mobile communication. The tremendous growth and usage of mobile network has enhanced the need for meeting better quality of service requirements. However, provisioning of QoS in wireless networks is complicated owing to the user mobility and limited network resources. Effective resource management is required to provide available resources during the call set up time as well as throughout the lifetime of connection. Call admission control (CAC) is a fundamental mechanism to ensure effective network resource management and to meet the QoS requirements of users. It involves making a decision for every new call request if there are enough idle resources to meet the QoS requirements without violating the QoS for already accepted calls.

Among various call admission control schemes, distributed call admission control (DCAC) consider both the local information (i.e. the amount of unused bandwidth in the cell where the user currently resides) and remote information (i.e. the amount of unused bandwidth in the neighboring cells) in order to determine whether to accept or reject the connection. It has been shown by various researchers that an admission control scheme that relies solely on local information cannot guarantee QoS requirements of a connection throughout its lifetime (Naghshineh and Schwartz, 1996; Wu et al.1998; Jiang et al, 2001;Kamble and Gupta, 2010).

Most of the researchers in distributed call admission control have dealt separately with the call admission control approaches in wireless networks. DCAC schemes proposed in the literature are mostly meant for only a single traffic type and the scheme may not work well if the network carries diverse types of traffic. In other schemes, call admission threshold is required to be estimated based on the assumption that all the admitted new calls are in progress at the beginning of control period. But in practice, some of the ongoing calls at the control period T may include the calls carried forward from the previous period and rest are new calls arriving at any time during the period. Other call admission control schemes for mobile networks are based on criterion like threshold level, handoff prioritization involving handoff, network status etc. (Yoon and Lee, 1999; Kim et al.2000). However these schemes involve the accurate measurement of threshold level.

In this study, an intelligent distributed call admission control scheme based on Fuzzy logic is proposed to ensure QoS for multimedia traffic applications. The proposed scheme is distributed in the sense that it takes into consideration of both local as well as remote information from neighboring cells and accordingly allocates bandwidth. The proposed scheme also takes into consideration of diverse traffic types and gives higher priority to real time calls compared to non-real time calls in order to provide higher connectivity to users. In addition to this, the performance of the proposed scheme is compared with that of simple guard channel scheme in terms of major QoS parameters such as new call blocking probability, call dropping probability and bandwidth utilization.

The paper is organized as follows. In section 2, related work in this area is described. Section 3 covers the proposed Fuzzy logic based distributed CAC (FDCAC) scheme in detail. In Section 4, simulation results are discussed and the performance of the proposed scheme is analyzed. Finally concluding remarks are presented in Section 5.

2. RELATED WORK

One of the earliest methods of ensuring call admission control in mobile networks is the guard channel scheme (Daigle and Jain, 1992; Oh and Tcha, 1992). In this scheme, each cell reserves a number of channels for exclusive use by handoff users. Various variants of this scheme exist based on the number of guard channels selected by the base station. Though the strategy is simple because there is no need for exchange of control information between the base stations, but wastage of scarce resources occur because of its poor adaptability to changing traffic load. In (Jiang et al.2001), a bandwidth reservation scheme is proposed, where a fixed number of channels are reserved exclusively for handoffs in each cell. The scheme also allows queuing of handoff requests when the reserved channels are not available in the cell. However, the proposed scheme does not consider information regarding the neighboring cells.

Several call admission algorithms using dynamic schemes have been proposed by various researchers (Zhang and Zhu, 2005; Kim et al.2000). The dynamic call admission schemes are based only on information gathered locally by each base station. In (Hou and Papavassiliou, 2001), a dynamic reservation based CAC scheme is proposed using the concept of influence curve. According to this scheme, a moving user exerts some influence on the channel allocation in neighboring cells. This is related to the mobility pattern (i.e. speed anddirection) of each specific user. However, it may not be practical always to characterize the random moving pattern of the users in wireless networks. Though these schemes provide scalable solutions but can lead to overestimation of resource requirements, thus adversely affecting the bandwidth utilization. In order to overcome these drawbacks, distributed call admission control (DCAC) algorithms that takes into consideration of the information exchanged from the neighboring cells have been proposed (Iraqi and Boutaba, 2000; Jiang et al.2001). However, in (Iraqi and Boutaba, 2000), only a single traffic type is considered and the scheme is not adaptive to changes in network conditions and it may not work well if the network carries diverse types of traffic.

In (Kim et al.2000), a modified DCAC scheme is described that does not follow the assumption of Poisson admission process. However, the proposed scheme requires frequent measurement of probability that a call hands off to other cells and probability for a call to remain in the same cell. Further the proposed DCAC is not applicable to multimedia environment. A distributed call admission algorithm has been proposed by (Ramanathan et al.1999) that guarantees upper bound of the cell overload probability. Also the authors used bandwidth adaptation algorithm that seeks to minimize the cell overload probability. (Zhang and Zhu, 2005) proposed distributed intelligent CAC scheme for wide band multi service CDMA system. Their results indicated that the proposed algorithm can guarantee the QoS requirements of users in terms of dropping probability and call blocking probability.

In (Naghshineh and Schwartz, 1996), DCAC scheme with aggregate resource reservation using mobility prediction based on mobile positioning system location information has been proposed and it takes into account the expected bandwidth to be used by calls handing off to and from neighboring cells within a configurable estimation time window. However the use of global positioning system for predicting user mobility leads to signaling overheads. According to DCAC scheme proposed in (Levine et al.1997), number of active calls present in both the local cell and its neighboring cells need to be known in order to determine its new call admission threshold. In addition to this, call admission threshold is estimated based on the assumption that all the admitted new calls are in progress at the beginning of control period. However, this assumption may cause imprecision in the control and makes the implementation of scheme difficult and thus leading to over provisioning of resources for handoff calls. Further, in this scheme, only a single traffic type is considered because ofwhich it may not work well if the network carries diverse type of traffic and is not adaptive to changes in network conditions.

Review of literature reveals that various DCAC schemes rely on unrealistic, simplifying approximations and assumptions, leading to imprecise control decisions. In addition to this, the computational complexity of the CAC algorithms based on Markov models becomes very high due to the number of states describing the system (Jayaram et al.2000). Fuzzy logic based control schemes have been investigated in various areas of traffic control such as CAC and congestion control (Naghshineh and Schwartz, 1996; Oliveira and Kim, 1998; Oh and Tcha, 1992) where they have demonstrated the ability to make intelligent control decisions successfully. Furthermore, Fuzzy logic can overcome most of the assumptions and complicated computations as the fuzzy if-then rules are based on linguistic variables that incorporate human-like knowledge representation of information.

3. PROPOSED FUZZY LOGIC BASED DCAC (FDCAC) MODEL

The objective of the proposed FDCAC scheme is to maximize the effective bandwidth utilization and to fulfill the QoS requirements in terms of achieving minimum handover dropping probability and new call blocking probability. Let us consider 2-D array of hexagonal cells as shown in Fig 1. Let λh be the handover call arrival rate into the cell, PB be the probability that a newly arriving call will be blocked. Maximum number of calls that can be supported in the cell be N_{max}.

Consider any test mobile in a radio cell C0 and let the number of calls in a cell at time to be k. It is assumed that the test mobile resides in the same cell with probability ps and that it hands-off to the adjacent cell with the probability of $p_m/6$, as the current cell is surrounded by six neighboring cells and an existing call within a cell may handover to any of the six cells. Thus, the values of p_s and p_m can be expressed as (Naghshineh and Schwartz, 1996, Baldo et al.1999)

$$p_m = [1-e^{-hT}] \text{ and } p_s = e^{-(h+\mu)T}$$

where T indicates the estimated time during which the call performs only one handover. According to this model, handover dropping probability is approximated to be equal to overload probability and is given as

$$P_{HD} \approx P_O = \sum_{i=N+1}^{\infty} Pi$$

where Pi is the probability that there are i active calls in a cell. Overload probability is the probability that the cell cannot support any more calls after all the channels available are used up.

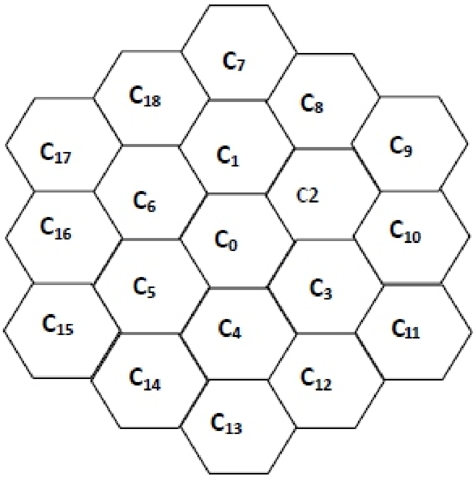

Fig. 1 Two dimensional cellular configuration

Therefore, from this approximation, handover dropping probability PHD can be evaluated as

$$P_{HD} \approx P_O = \sum_{i=N+1}^{n_1+n_2+n_3+n_4+n_5+n_6} P^0_{t+T}(i)$$

$$\approx Q\{N-m)/\sigma\}$$

t^2

Where $m = n_1 ps + (n_1+n_2+n_3+n_4+n_5+n_6)\, p_m/6$

$\sigma^2 = n_1 ps\,(1-ps) + (n_1+n_2+n_3+n_4+n_5+n_6) \bullet (1-p_m/6) \bullet p_m/6$

and $Q(x) = (1/\sqrt{2\Pi}\int e^{-t2/2}\,dt$, $x \geq 0$, Q(x) is an integral of tail distribution of standard normal distribution function.

Further, PB can be calculated by considering steady state probabilities of seven cells (i.e. the current cell and six neighboring cells) and it is given as

$$P_B = \sum_{n0=0}^{N_{max}} \sum_{n1=0}^{N_{max}} \sum_{n2=0}^{N_{max}} \sum_{n3=0}^{N_{max}} \sum_{n4=0}^{N_{max}} \sum_{n5=0}^{N_{max}} \sum_{n6=0}^{N_{max}} P(n_0)P(n_1)\,P(n_2)\,P(n_3)\,P(n_4)\,P(n_5)\,P(n_6)$$

where n_i denotes the number of existing calls at instant t and let $P(n_1)$, $P(n_2)$, $P(n_3)$, $P(n_4)$, $P(n_5)$, $P(n_6)$ be the steady state probabilities in the neighboring cell and N_{max} is the maximum possible number of calls in a cell.

In the proposed DCAC scheme, cells exchange their status information which is limited to nearest or next nearest neighboring cells. Thus in the present scheme, average handover dropping probability of the current cell and its adjacent neighboring cell (P_{ah}) is exchanged periodically. Furthermore, multiple channels can be allocated to a single user to satisfy higher bandwidth requirements of the user. It is assumed that when a new call or handoff call request is received at the base station, it provides the information regarding the type of the traffic class and the average bandwidth required for the connection. In the proposed model, different call admission criteria would be used for different class of traffic types and details are provided in the Table 1.

Table 1 Different categories of Traffic Classes considered in the study

Type of Traffic Class	Average Bandwidth requirement	Average connection duration (secs)	Examples
Class 2 (Real time) High*	30 Kbps	180	Voice and audio phone
	256 Kbps	300	Video phone and video conferencing
	3Mbps	600	Multimedia and video on demand
Class 1 (Non real time) Medium*	256 Kbps	180	Remote log in; data on demand
	5 Mbps	120	File transfer
Class 0 (Non real time) Low*	10 Kbps	30	Email and Fax, internet browsing
Source: Oliveira et al.,[20].			

* This is as per the membership function shown in Figure 3.

Call admission control algorithm is designed for both new call request and handoff calls. In the proposed algorithm, higher priority is assigned to real time handoff traffic (i.e. real time handoff and new call request).

For the present study, two variations of the proposed model are considered. For the first scheme, no bandwidth is reserved exclusively for high priority calls in the current or neighboring cells. For both the schemes, let the available bandwidth in the current cell be BA. When the network load in the cell is normal, the proposed scheme tries to allocate the average required bandwidth to every incoming call in the cell and let this bandwidth be denoted as b_{avg}. When a high priority real time handoff call comes, if the average required bandwidth is available in the current cell and average handover dropping probability in the neighboring cells and current cell (P_{ah}) is below the threshold level, i.e. PQoS, then the call is strongly accepted. If these conditions are not met, then the available bandwidth is compared with minimum required bandwidth. Thus if the available bandwidth at least satisfies the minimum required bandwidth and (P_{ah}) is below the PQoS level, then the call is accepted otherwise it is rejected. Similarly when real time new call request, non real time handoff call request or non real time new call request is received, if the available bandwidth exceeds the average bandwidth and average handover dropping probability is less than the PQoS, then the call is accepted, otherwise it is rejected. The pseudocode for this scheme is summarized as follows,

Pseudocode for FDCAC scheme A (without reservation)

If the incoming call is real time handoff request,

If ($B_A \geq b_{avg}$) and ($P_{ah} \leq P_{QoS}$), then

strongly accept the call request and assign average required bandwidth.

else if ($B_A \geq b_{min}$) and ($P_{ah} \leq P_{QoS}$), then

strongly accept the call and allocate the minimum required bandwidth.

else reject the call request

If the incoming call is one of the following cases:

(i) real time new call request

(ii) non real time handoff call request, or

(iii) non real time new call request , then the pseudocode is summarized as below,

If ($B_A \geq b_{avg}$) and ($P_{ah} \leq P_{QoS}$), then

strongly accept the call request and assign average required bandwidth.

else reject the call request.

In the second scheme, the user movement pattern is assumed to be unknown and some amount of bandwidth is reserved for high priority handoff real time calls in the current and all neighboring cells. The amount of bandwidth to reserve would depend on the number of existing high priority real time calls in the current cell and the minimum required bandwidth. Let this reserved bandwidth be represented as BR, then the total available bandwidth would be B_{TA} (= B_A + B_R). When real time handoff call is received, if the total available bandwidth is greater than the average required bandwidth and also average handover dropping probability is below the maximum defined value P_{QoS}, then the call is strongly accepted else the minimum required bandwidth is compared with available bandwidth as in previous case. When the real time new call request, non real time new call or non real time handoff call is and if the total available bandwidth is greater than the average required bandwidth and also average handover dropping probability is less than the P_{QoS}, then the call is accepted. However if this condition is not true, if the total available bandwidth exceeds the minimum required bandwidth and average handover dropping probability is less than the P_{QoS}, then the call is accepted else it is rejected. The details of this pseudocode is summarized as follows,

Pseudocode for FDCAC scheme B (with reservation)

If the incoming call is real time handoff request,

If $(B_{TA} \geq b_{avg})$ and $(P_{ah} \leq P_{QoS})$, then

strongly accept the call request and assign average required bandwidth.

else if $(B_{TA} \geq b_{min})$ and $(P_{ah} \leq P_{QoS})$, then

strongly accept the call and allocate the minimum required bandwidth.

else reject the call request

If the incoming call is one of the following cases:

(iv) real time new call request

(v) non real time handoff call request, or

(vi) non real time new call request , then the pseudocode is summarized as below,

If $(B_{TA} \geq b_{avg})$ and $(P_{ah} \leq P_{QoS})$, then

strongly accept the call request and assign average required bandwidth.

else reject the call request.

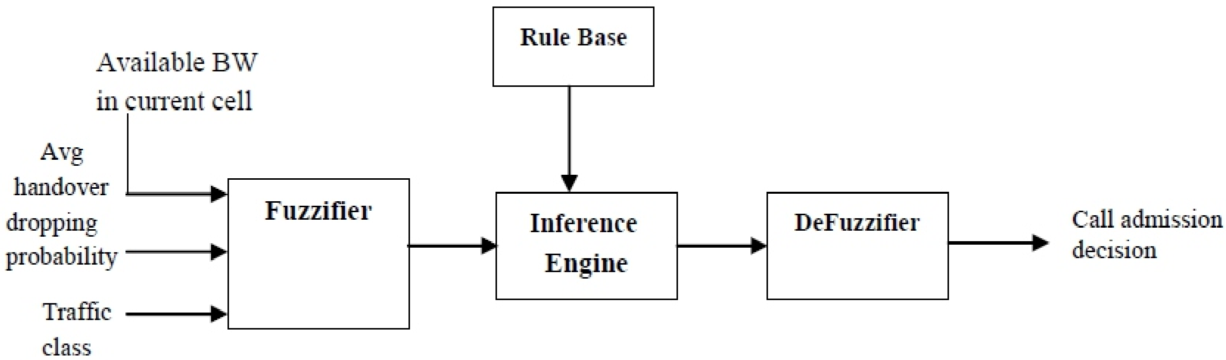

Fig.2 Fuzzy based Distributed call admission control Scheme

As can be seen, the controller comprises of Fuzzifier, Inference engine, Fuzzy rule base and Defuzzifier. All the real valued inputs are converted into corresponding linguistic variable values by Fuzzifier. The input parameters to the Fuzzifier unit include type of the traffic class that may be real time or non real time call requests; available bandwidth in current cell and average handover dropping probability in current and neighboring cells. Different types of traffic class are elaborated in Table 1. Real time services include voice and video services that are time and delay sensitive as well as loss sensitive. While services like data applications, email etc. that can tolerate large delays are classified as non real time applications. Real time handoff call requests have been given higher priority since in a multimedia networking environment; priority must be given to real time users because of limited resources (Baldo at al., 1999; Jayaram et al.,2000).

As can be seen from the Table 1, real time services have been classified as Class 2 (High) traffic services, similarly non real time traffic as Class 1 (medium) and Class 0 (low) for the Fuzzy rules. According to this, the membership function is depicted in Fig. 3. The most commonly used membership functions include Gaussian, triangle and trapezoid. For the present work, triangle functions are used. Membership functions for other inputs and outputs are depicted in Figs 3 to 6.

The membership function for average handover dropping probability (Baldo at al., 1999) is shown in Fig 5.

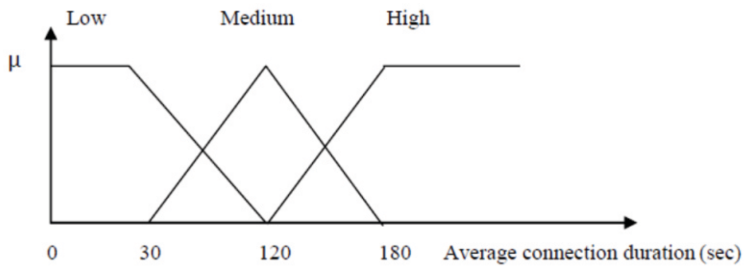

Fig. 3 Membership function for Traffic class

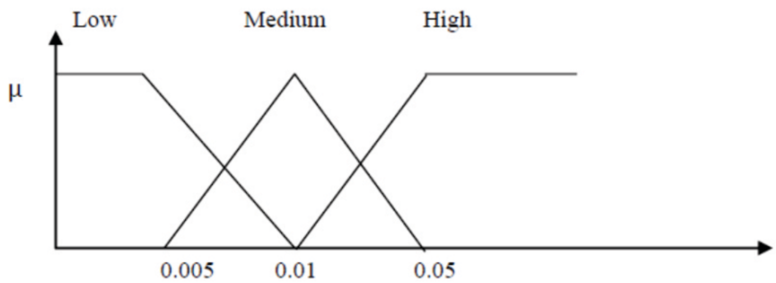

Fig. 4 Membership function for average HO dropping probability in current and neighboring cells

It is assumed that the maximum bandwidth capacity of a cell to be B Kbps.

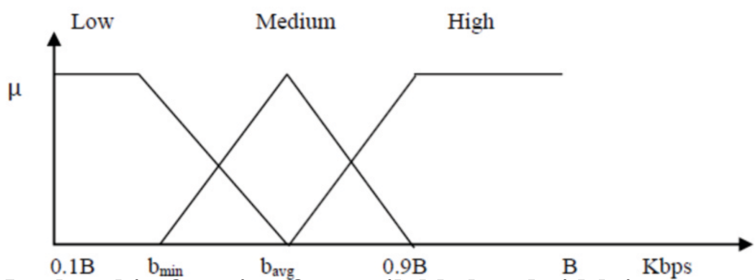

Fig. 5 Membership function for available bandwidth in current cell

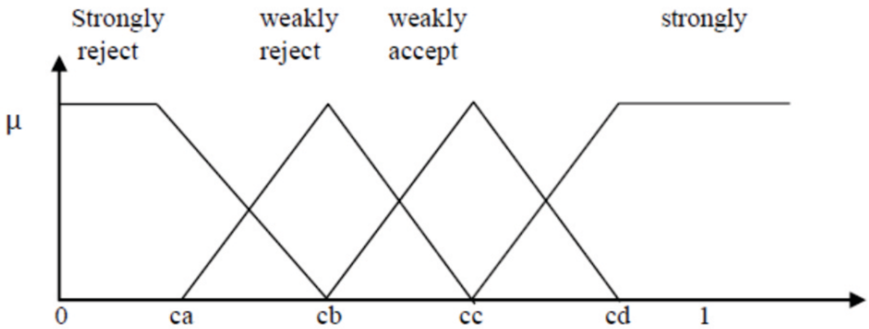

Fig. 6 Membership function for Call admission decision

The inputs to the FDCAC scheme are the mean values and are represented by their corresponding membership functions. Input fuzzy sets are labeled by their linguistic terms as low, medium and high. Fuzzy inference engine evaluates the set of If-Then rules which defines the system behavior. The result of this process is again a linguistic value, thus de- fuzzifiction step is required to set the output in discrete format. For this, centre of gravity method has been used as defuzzification method. The output i.e. the call admission decision for new call requests or handover calls is represented by the fuzzy sets as weakly accept (WA), strongly accept (SA), weakly reject (WR), strongly reject (SR). Some of the select Fuzzy rules for call admission control for handoff as well as new call requests are described in Table 2 and Table 3 respectively.

Table 2 Select Fuzzy rules for Handoff call request

IF			Then
Traffic class	**Average handover dropping probability in current and neighboring cell**	**Available BW in current cell**	**HO call admission decision**
Low	Low	Low	SR
Low	High	High	SR
Low	High	Medium	SR
Medium	Low	Medium	SA
Medium	High	High	SR
Medium	Medium	High	WA
High	Low	Medium	SA
High	Medium	Low	SR
High	High	Medium	SR

Table 3 Select Fuzzy rules for New call request

IF			Then
Traffic class	**Average handover dropping probability in current and neighboring cell**	**Available BW in current cell**	**call admission decision**
Low	Low	Low	WA
Low	High	High	SR
Low	High	Medium	WA
Medium	Low	Medium	SA
Medium	Medium	High	WA
Medium	High	High	SR
High	Medium	Low	SR
High	Low	Medium	WA
High	High	High	SR

The performances of the two proposed schemes have been compared with fixed reservation guard channel scheme in which fixed amount of guard channels are reserved exclusively for hand off calls only. The rest of the base station bandwidth is available for both new and handoff calls requests and priority is given to handoff calls. The details of the simulation are described in the following section.

4. SIMULATION RESULTS AND DISCUSSION

The simulation model comprises of cellular mobile network in which each hexagonal cell is assumed to be surrounded by six neighboring cells. A 2-D model of cellular system is considered that consists of 18 cells arranged in a circular pattern as shown in Figure 1. Each connection is assumed to experience multiple handoffs in its lifetime. It is assumed that each cell has a base station at the centre, which sets up and releases the connection (handoff and new call) and fixed channel allocation scheme is considered. It is assumed that the user while establishing the call also specifies the requirement profile in terms of average bandwidth required and specifies whether the call is real time or non real time. In addition to this, the new connections of various classes of traffic are generated with equal probability.

For the simulation purpose, the maximum bandwidth capacity of a cell is assumed to be 30 Mbps. For the present case, the performance of the proposed algorithm is compared with guard channel reservation model, thus fixed amount (5%) of the total bandwidth is reserved in the current cell exclusively for handoff calls in case of guard channel scheme. Each cell is offered same new originating traffic load, and the offered load to each cell is changed continuously. The above model was simulated in QualNet environment which is a discrete event network simulator and includes wide set of detailed models for wireless networking (QualNet user's manual). All simulations were repeated 20 times so that an averaged overall performance could be obtained and confidence interval was chosen to be 95%. The performance of two variants of Fuzzy logic based DCAC scheme is evaluated in terms of QoS parameters like blocking probability of new call requests, call dropping probability and bandwidth utilization. Fig. 7 compares the new call blocking probability for both the versions of FDCAC scheme and the guard channel scheme for class 2 traffic type.

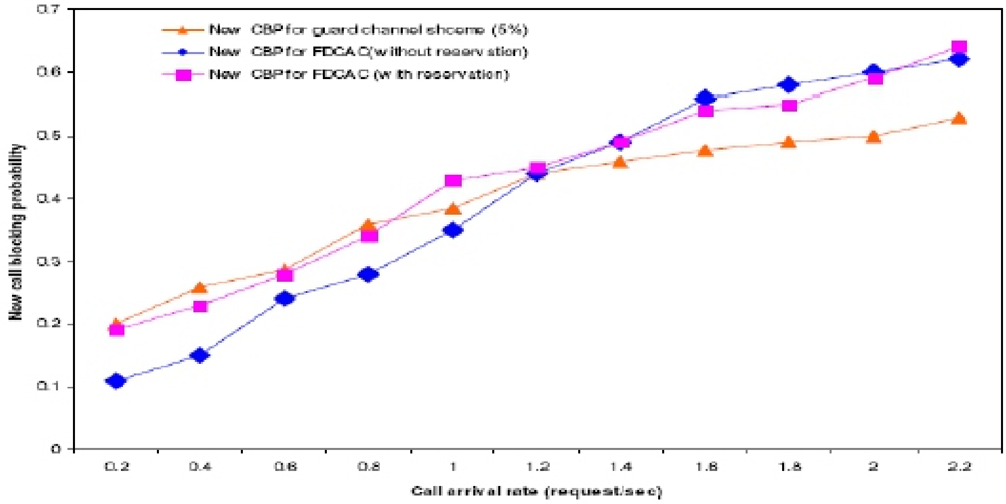

Fig. 7. New call blocking probability for Class 2 Traffic type

It can be seen that new call blocking probability for the guard channel scheme is lower compared to the FDCAC scheme, except under low load conditions. But with the further increase in the load, FDCAC without reservation scheme provides higher call blocking probability. Similarly, call dropping probability of both the FDCAC schemes is compared with the guard channel scheme for class 2 traffic type as depicted in Fig. 8.

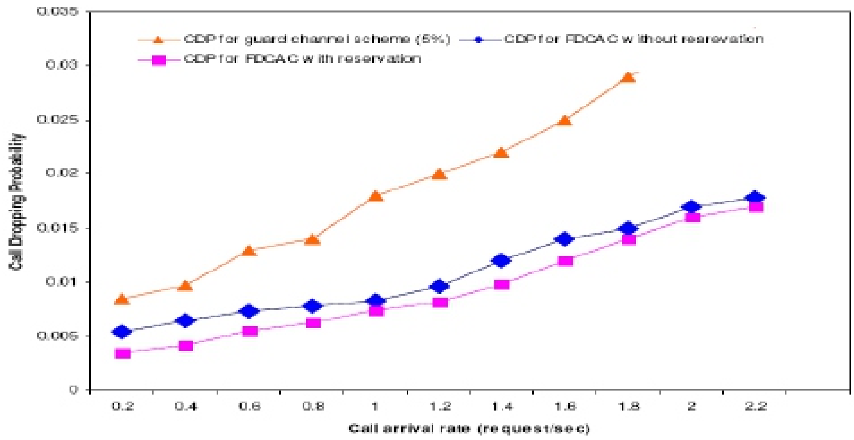

Fig. 8. Call dropping probability for Class 2 Traffic type

Fig. 8 shows that call dropping probability in case of FDCAC with reservation is less compared to the guard channel or FDCAC without reservation scheme. However as can be seen that the difference in CDP values for these two versions of FDCAC is not very significant.

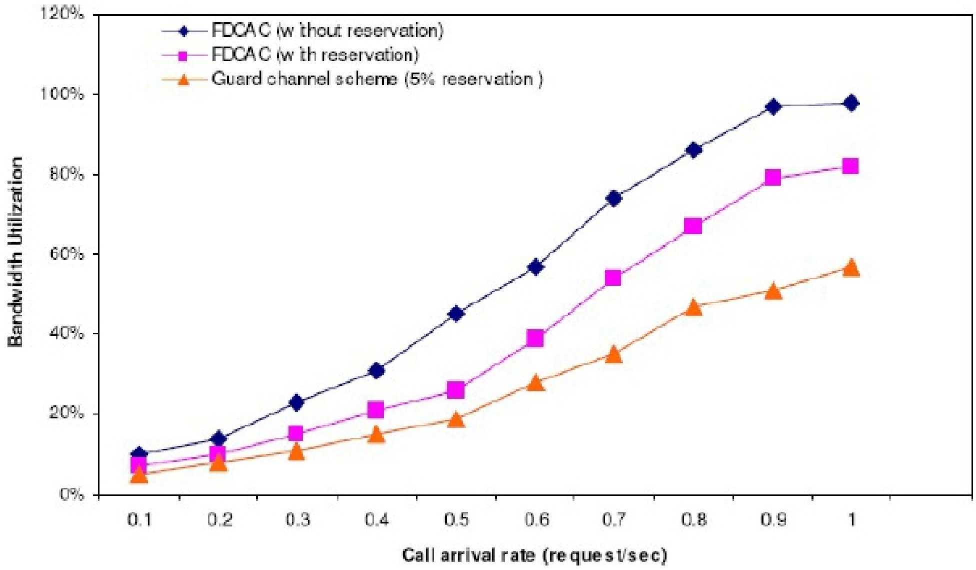

Fig. 9. Comparison of Bandwidth Utilization for the schemes

Fig. 9 depicts the bandwidth utilization for the guard channel and both the FDCAC schemes under study as a function of call arrival rate. The Figure indicates that the bandwidth utilization of both the variants of FDCAC scheme gradually increases with call arrival rate and is higher compared to guard channel scheme. However, FDCAC without reservation provides better utilization of bandwidth. From these results, it is evident that proposed FDCAC scheme without reservation fulfils most of the QoS requirements. The difference in the performance in terms of both the new call blocking probability and call dropping probability between the two variants is less bandwidth utilization of FDCAC without reservation is better. So it would be quite useful to focus on FDCAC without reservation and the cost of QoS guarantees need to be identified for the scheme B.

5. CONCLUSIONS AND FUTURE WORK

In this study, an intelligent distributed call admission control scheme using Fuzzy logic has been proposed to ensure QoS in multimedia environment. Two variations of fuzzy logic based FDCAC scheme have been considered. In the first scheme, no bandwidth is reserved exclusively for high priority calls in the current or neighboring cells. In the second scheme, the user movement pattern is assumed to be unknown and some amount of bandwidth is reserved for high priority handoff real time calls in the current and all neighboring cells. The performance of these two fuzzy logic based schemes has been compared with guard channel scheme.

It is shown through simulations that the performance of both FDCAC schemes is better compared to guard channel scheme in terms of QoS parameters considered. Further, the call dropping probability for FDCAC with reservation (scheme B) is not showing significant improvement compared to FDCAC without reservation (scheme A) scheme. It is also revealed that FDCAC scheme without reservation achieves better bandwidth utilization compared to other two schemes. This indicates that the performance of FDCAC scheme without reservation is better than the other two schemes. The work can be extended to take into account of different channel assignment techniques and also non-uniform traffic pattern can be considered in future work.

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Survey: Shortest Path Routing over Mobile AD HOC Networks

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ABSTRACT

Over the past decade or so, there has been rapid growth in wireless and mobile applications technologies. More recently, an increasing emphasis has been on the potential of infrastructure less wireless mobile networks that are easy, fast and in-expensive to set up, with the view that such technologies will enable numerous new applications in a wide range of areas. Such networks are commonly referred to as mobile ad hoc networks. In MANET, data is sent from one node to another node by using links, called path. Routing protocol is responsible to find out the path to be followed by the packets from source to destination. For MANET, there are different type of protocols are available and as per the requirement, we can use any one. Each protocol has its own criteria to find the path from source to destination. Some of researchers explored the concept of shortest path routing over ad hoc network. In this paper, we will discuss the solution given by them.

Keywords: Ad-hoc network, Bellman Ford, Dijkstra, MANET, Routing Protocol, Shortest Path

1. INTRODUCTION

Ad hoc network consists of autonomous devices that can directly communicate to their nearby nodes. Nodes that are not within direct communication range use other nodes to relay messages between them. Routing in such an ad hoc network is challenging due to the lack of central control and the high dynamics of the network. Nodes are free to move in any direction and organize themselves arbitrarily. They can join or leave the network at any time. Due to the dynamic network topology there is a frequent change in the status of routing table which adds the complexity to routing among the various mobile nodes. Routing protocols for MANETs are usually classified into proactive and reactive protocols, and hybrid protocols based on how routing information is acquired and maintained by mobile nodes. Table proactive protocols use a proactive Routing scheme, in which every network node maintains consistent up-to-date routing information from each node to all other nodes in the network. On-demand-reactive protocols are based on a reactive routing scheme, in which at least one route is established only when needed. A hybrid routing protocol is a combination of proactive and reactive schemes with the aim of exploiting the advantages of both types of protocols. [1][12]

Shortest Path algorithm

There are two basic shortest path algorithms are available i.e. Bellman ford algorithm and Dijkstra's algorithm for shortest path problem. We can use one of them to solve the routing problem. Routing based on these algorithms performs the following steps:

1. Each node calculates the distances between itself and all other nodes within the network and stores this information as a table.
2. Each node sends its table to all neighbouring nodes.

When a node receives distance tables from its neighbours, it calculates the shortest routes to all other nodes and updates its own table to reflect any changes. Destination Sequenced Distance Vector (DSDV) routing protocol has been developed for ad hoc network. It is based upon the distributed Bellman-Ford algorithm. [1][13]

2. LITERATURE SURVEY

Mobile ad hoc network (MANET) is a self-organizing and self-configuring multi-hop wireless network, which is composed of a set of mobile hosts (MHs) that can move around freely and cooperate in relaying packets on behalf of one another. MANET supports robust andefficient operations by incorporating the routing functionality into MHs. In MANETs, the unicast routing establishes a multi-hop forwarding path for two nodes beyond the direct wireless communication range. Routing protocols also maintain connectivity when links on these paths break due to effects such as node movement, battery drainage, radio propagation, and wireless interference. In multi-hop networks, routing is one of the most important issues that have a significant impact on the performance of networks. [1]

Concept of shortest path can be used with any routing algorithm but the result will depend upon the nature of the protocol and the parameters used. Lots of research work has been done using shortest path routing over wireless ad hoc network with the different constraints and with different protocols. Now we will discuss different shortest path routing schemes by the researchers.

W. Ahn and R. S. Ramakrishna [2] proposed a genetic algorithmic approach to the shortest path (SP) routing problem. Variable-length chromosomes (strings) and their genes (parameters) have been used for encoding the problem. The crossover operation exchanges partial chromosomes (partial routes) at positional independent crossing sites and the mutation operation maintains the genetic diversity of the population.

The proposed algorithm can cure all the infeasible chromosomes with a simple repair function. Crossover and mutation together provide a search capability that results in improved quality of solution and enhanced rate of convergence. They also developed a population-sizing equation that facilitates a solution with desired quality. It is based on the gambler's ruin model; the equation has been further enhanced and generalized, however. The equation relates the size of the population, the quality of solution, the cardinality of the alphabet, and other parameters of the proposed algorithm. Computer simulations show that the proposed algorithm exhibits a much better quality of solution (route optimality) and a much higher rate of convergence than other algorithms.

A. C. Valera and K.G. Seah [3] proposed a new routing protocol named CHAMP (Caching and Multiple Path) routing protocol. CHAMP uses cooperative packet caching and shortest multipath routing to reduce packet loss due to frequent route failures. Extensive simulation results that these two techniques yield significant improvement in terms of packet delivery, end-to-end delay and routing overhead. It was proposed that existing protocol optimizations employed to reduce packet loss due to frequent route failures, namely localrepair in AODV and packet salvaging in DSR, are not effective at high mobility rates and high network traffic.

Y. Shavitt and A. Shay [4] introduced the Gossip Network model where travellers can obtain information about the state of dynamic networks by gossiping with peer travellers using ad hoc communication. Travellers then use the gossip information to recourse their path and find the shortest path to their destination. They studied optimal routing in stochastic, time- independent gossip networks, and demonstrate that an optimal routing policy may direct travellers to make detours to gather information.

A dynamic programming equation that produces the optimal policy for routing in gossip networks is presented. In general, the dynamic programming algorithm is intractable; however, for two special cases a polynomial optimal solution is presented. Results show that ordinarily gossiping helps travellers decrease their expected path cost. However, in some scenarios, depending on the network parameters, gossiping could increase the expected path cost. The parameters that determine the effect of gossiping on the path costs are identified and their influence is analyzed. This dependency is fairly complex and was confirmed numerically on grid networks.

S. Jiang and J. Rao [5] introduced a prediction-based link availability estimation to quantify the link reliability. This quantity makes use of some instantly available information and also considers the dynamic nature of link status in order to properly reflect the link reliability. Then, this quantity has been further used to develop routing metrics for path selection in terms of path reliability to improve routing performances. The proposed schemes have been investigated through computer simulation.

J. Gao and Li Zhang [6] studied routing algorithms on wireless networks that use only short paths, for minimizing latency, and achieve good load balance, for balancing the energy use. They considered the special case when all the nodes are located in a narrow strip with width at most pffi3ffiffi=2_0:86 times the communication radius. They presented algorithms that achieve good performance in terms of both measures simultaneously.

In particular, the routing path is at most four times the shortest path length and the maximum load on any node is at most three times that of the most load-balanced algorithm without path-length constraint. In addition, our routing algorithms make routing decisions by only local information and, as a consequence, are more adaptive to topology changes due to dynamic node insertions/deletions or due to mobility.

A. Arora and H. Zhang [7] formulated a notion of local stabilization, by which a system self- stabilizes in time proportional to the size of any perturbation that changes the network topology or the state of nodes. The notion implies that the part of the network involved in the stabilization includes at most the nodes whose distance from the perturbed nodes is proportional to the perturbation size. Also, they presented LSRP, a protocol for local stabilization in shortest path routing. LSRP achieves local stabilization via two techniques. First, it layers system computation into three diffusing waves each having a different propagation speed, i.e., “stabilization wave” with the lowest speed, “containment wave” with intermediate speed, and “super-containment wave” with the highest speed. The containment wave contains the mistakenly initiated stabilization wave, the super- containment wave contains the mistakenly initiated containment wave, and the super- containment wave self-stabilizes itself locally.

Second, LSRP avoids forming loops during stabilization, and it removes all transient loops within small constant time. To the best of our knowledge, LSRP is the first protocol that achieves local stabilization in shortest path routing.

Y. Shen, Y. Cai, X. Li, and X. Xu [8] presented an energy-efficient topology control algorithm named RLSP. The algorithm first tries to preserve the minimum-energy paths. However, when a node finds it needs a large transmission power to cover some of its logical neighbours, it uses two-hop paths to reach them instead of using single links. Simulation results show that RLSP can effectively decrease the transmission power and reduce the energy consumption when transmitting.

V. Lenders, M. May and B. Plattner [9] introduced density based any cast routing, a new any cast routing paradigm particularly suitable for wireless ad hoc networks. Instead of routing packets merely on proximity information to the closest member, density-based any cast routing considers the number of available any cast group members for its routing decision. They present a unified model based on potential fields that allows for instantiation of pure proximity-based, pure density-based, as well as hybrid routing strategies.

They implemented any cast using this model and simulate the performance of the different approaches for mobile as well as static ad hoc networks with frequent link failures. Ourresults show that the best performance lies in a trade-off between proximity and density. In this combined routing strategy, the packet delivery ratio is considerably higher and the path length remains almost as low than with traditional shortest-path any cast routing.

H. Dubois-Ferriere, M. Grossglauser, M. Vetterli [10] introduced opportunistic routing protocols by using single-path routing metrics to assign to each node a group of candidate relays for a particular destination. This paper addresses the least-cost any path routing (LCAR) problem: how to assign a set of candidate relays at each node for a given destination such that the expected cost of forwarding a packet to the destination is minimized. The key is the following trade-off: On one hand, increasing the number of candidate relays decreases the forwarding cost, but on the other, it increases the likelihood of “veering” away from the shortest-path. Simulations show significant reductions in transmission cost to opportunistic routing using single-path metrics.

L. Ying, S. Shakkottai, A. Reddy, and S. Liu [11] proposed a new routing/scheduling back- pressure algorithm that not only guarantees network stability (throughput optimality), but also adaptively selects a set of optimal routes based on shortest-path information in order to minimize average path lengths between each source and destination pair.

Results indicate that under the traditional back-pressure algorithm, the end-to-end packet delay first decreases and then increases as a function of the network load (arrival rate). This surprising low-load behaviour is explained due to the fact that the traditional back-pressure algorithm exploits all paths (including very long ones) even when the traffic load is light. On the other-hand, the proposed algorithm adaptively selects a set of routes according to the traffic load so that long paths are used only when necessary, thus resulting in much smaller end-to-end packet delays as compared to the traditional back-pressure algorithm.

3. PROBLEM FORMULATION

Mobile Ad hoc Network (MANET) is a collection of wireless mobile terminals that are able to dynamically form a temporary network without any aid from fixed infrastructure or centralized administration. One critical issue for routing in MANETs is how to select reliable paths that can last as long as possible since radio links may be broken frequently. The reliability of a path depends on the number of links and the reliability of each link constituting the path. Many routing metrics in terms of number of links have been proposed, such as the shortest path routing.

Shortest path routing selects a path having minimum cost to forward the data to next node. Shortest path selection may be done on the basis of different parameters like transmission cost which can be calculated on the basis of routing table information, link stability factor, power consumption factor etc. Performance of the network can be enhanced through shortest path routing but it also depends upon the functionality of the routing protocol and the parameters selected for the shortest path routing.

4. CONCLUSION

In this paper, we presented a brief survey on shortest path routing. Various solutions have been offered by different researchers and each researcher has used a different parameter for the shortest path for routing. Our study shows that result of the shortest path routing also depends upon the selected parameters and as well as on the selected protocol used for routing. Each author worked on a specific parameter but no one has considered all the parameters in a particular solution, discussed in this paper.

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