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Effects of routing algorithms on novel throughput improvement of mobile ad hoc networks

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Abstract

A cognitive method called most congested access first (MCAF), minimizing the packet loss ratio and improving the throughput of a multihop mobile WiMAX network, is proposed. MCAF combines both the time division multiple access and the orthogonal frequency division multiple access methods. MCAF additionally uses spectral aid and buffer management methods, which are proposed in this paper, to manage both spectrum access and packets in the buffers. By using these novel methods, real-time video and voice packet transmission is achieved, data packet loss rate is minimized, and the system throughput per node is improved. Effects of fastest path and ant colony routing algorithms on throughput improvement methods are investigated. It is shown that the fastest path routing algorithm provides higher throughput values than the ant colony routing algorithm.

Key Words: *Throughput, routing, cognitive, 802.16j, multimedia*

1. Introduction

The idea of cognitive radio (CR) was first presented in [1], where a better way of manipulating protocol stacks by defining radio knowledge representation language (RKRL) was proposed. RKRL was designed to be used by software agents with a higher level of competence, driven in part by a large storage of prior knowledge that may be of a cognitive nature. Mitola gave a description of cognitivity later in [2]. The introduction of cognitivity led to new challenges for the resource allocation and design of WiMAX relay-based systems.

Most works in the literature attempted to improve system throughput with the cooperation of primary and secondary users for efficient resource allocation [3,4]. However, there is very little work on the network throughput of multihop 802.16j networks [5]. In order to maximize throughput performance, the authors in [3] proposed a method for flexible channel cooperation, allowing secondary users to freely optimize the use of channels for transmitting their own data along with primary data.

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In [4], the authors focused on determining the throughput potential of CR for various transmission power levels of the secondary nodes and determining the optimal amount of licensing. However, design of a WiMAX relay-based system and per user throughput improvement was not considered in [3] and [4].

In [5], a study of a transparent mode relay-based 802.16j system performance was described by considering the design of WiMAX relay-based systems. However, only 5% throughput improvement was provided with almost twice the signaling overhead. In comparison, our method provides higher throughput values (up to 36%).

In [6], the authors addressed the problem of assigning channels to CR transmissions, assuming one transceiver per CR. They attempted to maximize the number of simultaneous CR transmissions. By decreasing the blocking rates of CR transmissions, 50% throughput improvement was provided for single-hop scenarios, but only 20% improvement was provided for multihop scenarios.

In [7], the throughput improvement of an 802.16j network was provided for a fixed number of nodes (N = 6). In [8], simulation results were generalized for arbitrary N values, and the simulation results for a traditional pure system (none of the proposed method is in use) were shown to match the results of [9].

In this paper, we use spectral aid (SA) and buffer management (BM) methods with the proposed most congested access first (MCAF) method to manage packets in buffers and provide effective spectrum sharing in a fair and cooperative way. The throughput of a traditional 802.16j network is evaluated initially for a fixed N value (N = 6) as in [7], and then extended to arbitrary N values. It is shown that real-time packet transmission is achieved, the loss rate of nonreal-time data packets is minimized, and system throughput is improved with each method. Finally, the effects of fastest path and ant colony routing algorithms on throughput improvement are investigated. The proposed methods are shown to lead to throughput improvements in both routing algorithms. The amount of spectral usage is also calculated with and without the bandwidth wastage. The throughput is then calculated for the pure system and compared with those of 3 different works [9-11] that provide the throughput of a unicast system by asymptotic analysis or by simulating the conventional relaying network. The results of probabilistic throughput calculations and simulations for a traditional 802.16j network are confirmed by the theoretical and simulation results reported in the literature.

To the best of our knowledge, this is the first analytically confirmed event-driven simulation work for WiMAX relay-based network design that focuses on decreasing the packet loss ratio, improving the throughput per user in a cognitive multimedia network, and investigating the effects of routing algorithms on throughput improvement.

2. Throughput of mobile ad hoc networks

2.1. Throughput analysis

The asymptotic throughput per user of a unicast system is given by [9]:

$$\begin{aligned} R_u &= \frac{B}{N}\log_2\left(1+\frac{\rho_0 \ln(N)}{d_c^n}\right)+\frac{Bn}{2\ln(2)N} \\ \rho_0 &= \frac{P}{N_0 \times B \times K} \times \beta. \end{aligned} \tag{1}$$

Here, B is the used bandwidth, N is the number of active nodes, d_c is the cell diameter, n is the path loss exponent, P/N_0 is the signal-to-noise ratio (SNR), K is the channel model constant, and β is the bit error rate (BER)-related value.

The corresponding formula from [10] may be written as:

$$R_u = \frac{B}{N} \times \log_2 \left(1 + \beta \times E\left\{\Gamma_{eff}\right\}\right), \tag{2}$$

where Γ_{eff} is the average effective SNR and takes the value of P/N$_0$ when the results of a system are being evaluated.

The throughput of conventional relaying used in the simulation study in [11] (one relay can transmit at a time) is given by:

$$BR = \frac{R\_OFDM(RS)}{FL\left(\sum_{i=1}^{m} \frac{SSG1_i}{bps_i} + \sum_{i=1}^{m} \frac{SSG2_i}{bps_i} + \frac{1}{bps}\sum_{i=1}^{m} SSG2_i\right)}, \tag{3}$$

where FL is frame length, BR denotes the nominal bit rate (bits/s), SSG is the number of nodes using the individual modulation type, bps is the number of bits that can be allocated to one OFDM symbol, and R_OFDM(RS) is the number of OFDM symbols needed. The parameter values of Eq. (3) are determined during the simulation according to the network state.

2.2. Comparative analysis of throughput results

The simulation results of [9-11] for the conventional 802.16j system were compared with each other using the same parameters used in [9], our simulation system, and [11]. Figure 1 shows that the simulation results of all works are consistent.

Figure 1a shows the simulation results for [9-11] using the parameter set in [9], where B = 1 MHz, FL = 5 ms (typical), P/N$_0$ = 10^3, d$_c$ = 1000 m, n = 3.5, K = $10^{3.15}$ (suburban NLOS channel model), and β, which is related to the BER, is$\beta = -1.5/\ln(5\,BER) = 0.2$.

The results illustrated in Figure 1b were obtained using our parameter set, where B = 10 MHz, FL = 5 ms (typical), P/N$_0$ = 10^3, d$_c$ = 50 m, n = 2 (free space), K = 10^0 (0 dB), and β = 0.02.

The results in Figure 1c were obtained by using the parameters from [11], where B = 20 MHz, FL = 20 ms, P/N$_0$ = 125.89 (21 dB for 64 QAM and FEC = 3/4), d$_c$ = 1000 m, n = 2 (free space), K = $10^{3.15}$ (suburban NLOS channel model), and β = 0.155 (obtained from the simulation).

Note that efficient spectral usage amount (or capacity) is defined as the amount of data successfully forwarded to its next node per second, and throughput is defined as the amount of data that has successfully arrived at its final destination per second.

Since Eqs. (1) and (2) determine single-hop capacity and Eq. (3) determines multihop throughput, the values obtained from both Eqs. (1) and (2) are divided for the average hop count (AHC) of 2.25 hops (obtained by the simulation for the scenario given in Eq. (3)) when comparing the results of [9] and [10] with the results of [11].

3. Simulation program

An event-driven simulation program in MATLAB was developed for this study, in which the movements, locations, and buffer states of N nodes; the organization and selected routes of the packets in the buffer of

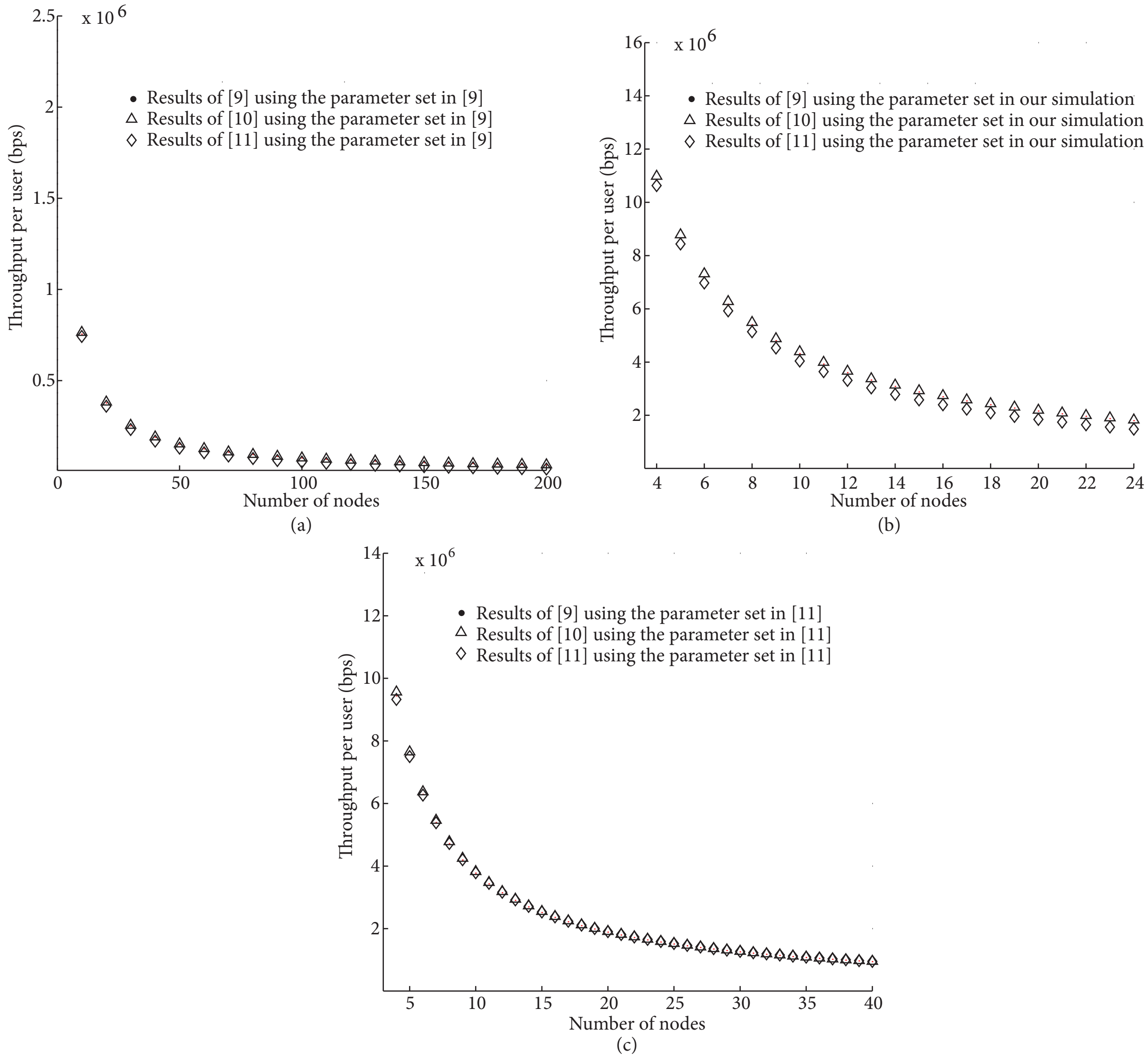

Figure 1. Comparative analysis of throughput results.

each node; instant data generation rates; and instant overall throughput values are all observable from the screen.

The overall algorithm used in the simulation is given in Figure 2. In the simulation, the relay nodes are considered to communicate with each other in a cell structure and the simulation parameter values can be changed to any desired value.

3.1. Determination of the maximum spectral usage

Before focusing on the purpose of this work, which is maximizing the network throughput, it must be clearly understood how the system calculates maximum spectral usage (MSU) and spectral usage amount (SUA) with and without bandwidth wastage and how it decides the packet sizes and buffer sizes. In the simulation, the

MSU provided by the system is determined using the parameter values defined by standards [12,13], where 64 QAM is used with a FL of 5 ms and the forward error correction (FEC) rate is taken as 3/4. Forty-four data symbols per frame (DSPF), 30 subchannels, and 10 MHz of bandwidth (B) were used in the simulation.

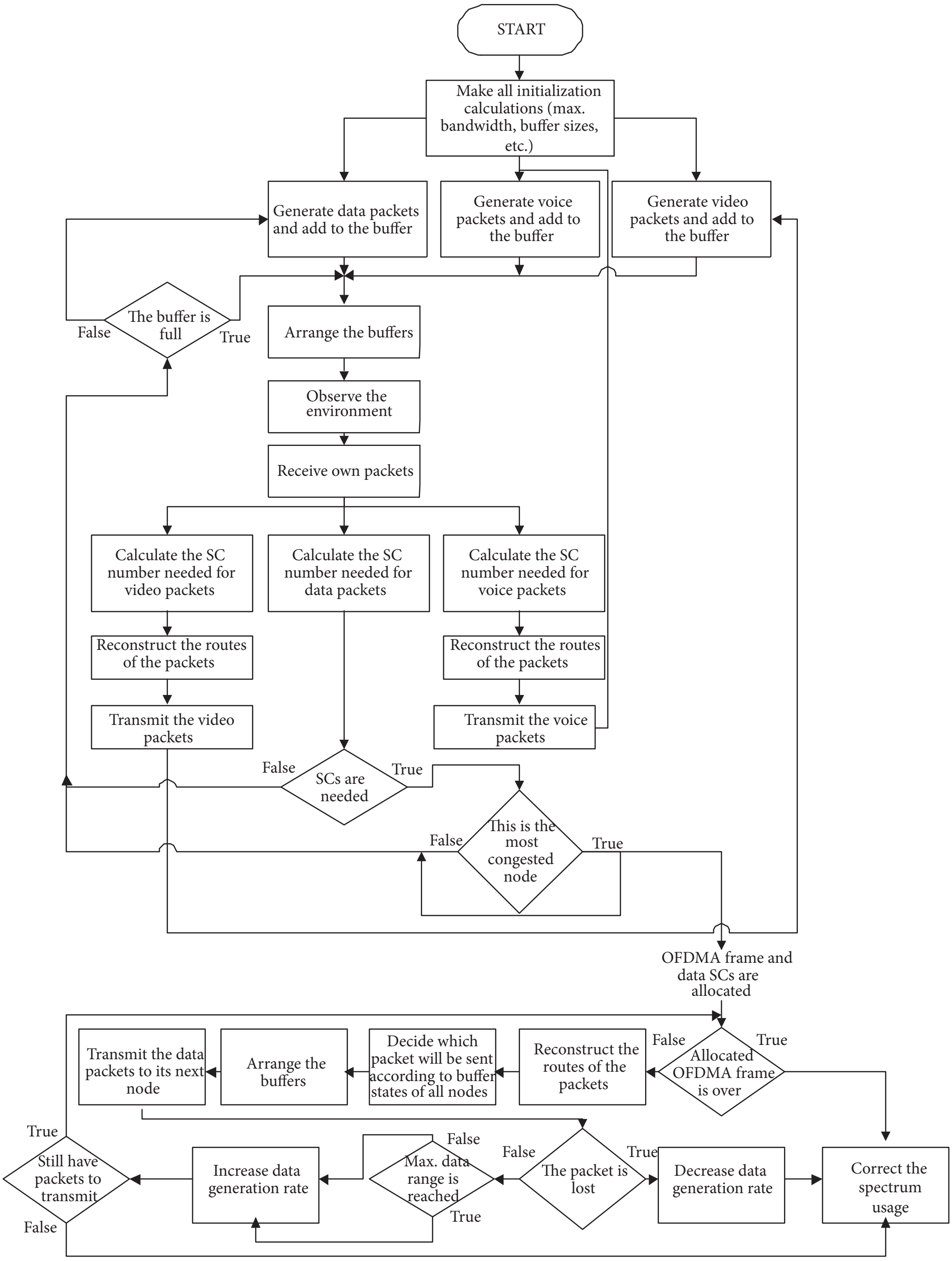

Figure 2. The overall algorithm used by each node in the simulation.

By using the given system parameters, the number of frames per second (FPS) is calculated by 1/FL and the number of bits per symbol is calculated by:

$$\text{Bits per symbol } (bps) = \log_2 (QAM) \times NODS, \tag{4}$$

where NODS is the number of data subcarriers, equal to 720 in the simulation. If FEC is used, the bps value will be multiplied by the used FEC rate and the number of bits per symbol with FEC (BPSWF) will be evaluated. The minimum allocatable unit (MAU) will be calculated by dividing the resultant BPSWF value by the number of subchannels (NOS). The subchannel data rate in a frame with FEC (SCDRFWF) will be evaluated by multiplying the MAU by the DSPF. Finally, the symbol rate per second (SRPS) will be evaluated by multiplying the DSPF by the FPS.

After doing these calculations with the given parameter values, SRPS can be evaluated as 8800 data symbols per second. The subchannel capacity with FEC (SCC) can be evaluated as 0.95 Mbps by multiplying the SRPS by the evaluated MAU value. Finally, the MSU can be evaluated as 3.564 Mbps, as in [12,13], by multiplication of the SCC by the NOS.

3.2. Determination of spectral usage amount

If the capacity of one subchannel is not a multiple of the used packet size, some parts of the subchannels in the spectrum cannot be completely filled. This causes wastage of the bandwidth. This problem is solved in our system by adjusting the packet size such that multiples of it completely fit in a subchannel. However, if a collection of generated small-sized packets (such as voice packets) cannot completely fill the subchannel, the unfilled part of the subchannel will again be wasted. Therefore, the calculated SUA value will differ when this bandwidth wastage is not taken into account.

3.2.1. Spectral usage amount with bandwidth wastage

Since the maximum possible hop count (MHC) for N nodes can only be N – 1 hops, a packet in the network may stay in the network for a maximum duration of N – 1 frames. For N × (N – 1) packet groups (generated by N nodes in the last N – 1 frames) multiplied by the voice packet sending rate (VPSR$_{voice}$ packets / frame) plus N × VPSR$_{voice}$ currently generated voice packets by N nodes, a total of [N + (N – 1) × N] × VPSR$_{voice}$ voice packets will be sent by all nodes in each frame. A total of [N + (N – 1) × N] × VPSR$_{voice}$ × VCPS (voice packet size) bytes of voice packets will then be transmitted by N nodes via N subchannels. This means that [N subchannels × MSU / NOS] – {[N + (N – 1) × N] × VPSR$_{voice}$× VCPS} bytes of subchannels will not be filled and will be wasted in a frame. Thus, the bandwidth wastage in 1 s for voice packets in N nodes (Wastage$_{Nvoice}$) is calculated by:

$$Wastage_{N_{voice}} = \left((N \times \frac{MSU}{NOS \times \frac{1}{FL}}) - (VCPS \times (N + (N-1) \times N) \times VPSR_{voice}) \right) \times \frac{1}{FL} Bps, \tag{5}$$

and SUA$_{with_wastage}$ is calculated by:

$$SUA_{with\_wastage} = MSU - Wastage_{N_{voice}} Bps. \tag{6}$$

3.2.2. Spectral usage amount without bandwidth wastage

If the spectrum is considered to be fully used without any bandwidth wastage, the SUA values will only differ due to the effects of data generation rates, the packet loss rates, and the AHC provided by the used routing algorithm, but not by the unused parts of subchannels. The SUA without bandwidth wastage can be evaluated as:

$$SUA_{without\_wastage_N} = AHC_N \times 8 \times \left[THR_{N_{with\_wastage}} + (wastage_{N_{voice}})\right] \div Nbps. \tag{7}$$

Note that the term $\mathrm{THR}_{\mathrm{N\_with\_wastage}}$ in Eq. (7) is the simulation throughput result from the source to the final destination and the term $\mathrm{wastage}_{\mathrm{N_{voice}}}$ is added to $\mathrm{THR}_{\mathrm{N\_with\_wastage}}$ assuming that the wasted bandwidth is also used for packet transmission and is not wasted. The results obtained from Eq. (7) can be confirmed by calculating the term $\mathrm{wastage}_{\mathrm{N_{voice}}}$ in Eq. (5), by using 1,225,488 bytes of pure system $\mathrm{THR}_{\mathrm{N_{with\_wastage}}}$ simulation results for N = 6, and by using the AHCs for N = 6 as $\mathrm{AHC}_6 = 2.96 \approx 3$ hops (obtained from the simulation using the fastest path routing algorithm). The result of Eq. (7) can be evaluated as 7,465,152 bps, and this value is very close to the corresponding simulation result, which is 7,638,500 bps. Since the bandwidth wastage is not taken into account in [9-11], we use $\mathrm{SUA}_{\mathrm{without\_wastage}}$ for confirmation purposes.

3.3. Calculation of the packet sizes of multihop mobile WiMAX network

For no bandwidth wastage, both real-time and nonreal-time packet sizes must be adjusted carefully such that multiples of them exactly fit into the subchannels.

3.3.1. Calculation of real-time multimedia packet sizes

The video packet size (VDPS) is evaluated at 594 bytes by using Eq. (11), substituting the term VDPS_{ref} in place of DTPS_{ref}. In the process, 512 bytes of reference video packet size (VDPS_{ref}) are used. Thus, VDPS exactly fits the SCDRFWF, which is obtained by:

$$SCDRFWF = \frac{\frac{MSU}{NOS}}{\frac{1}{FL}} = \frac{\frac{3564000\ bytes/s}{30SC}}{200fps} = 594\ bytes. \tag{8}$$

According to [12], 16-kbps voice packets can be considered due to low latency requirements, and the VCPS is calculated as:

$$VCPS = 16\ kbits \div 200 = 10\ bytes, \tag{9}$$

for 200 frames in 1 s with the frame length of 5 ms used in our system.

3.3.2. Calculation of nonreal-time data packet sizes

The number of subchannels in a frame not used by video or voice packets and allocated for nonreal-time data transmission can be calculated as:

$$TSCFDT = (NOS - (TSCFVD + TSCFVC)), \tag{10}$$

where TSCFDT, TSCFVD, and TSCFVC express the total number of subchannels allocated for data, video, and voice packets, respectively. Taking the reference data packet size (DTPS_{ref}) as 150 bytes, the chosen data packet size (DTPS) closest to DTPS_{ref} is calculated as:

$$\begin{aligned} DTPS &= \frac{(SCDRFWF)}{floor\left(\frac{SCDRFWF}{DTPS_{ref}}\right)} \\ &= \frac{594}{floor\left(\frac{594}{150}\right)} = 198\ bytes \end{aligned} \quad . \tag{11}$$

Here, SCDRFWF was calculated in Eq. (8) for NOS = 30 and FPS = 200.

3.4. Calculation of buffer sizes

Choosing a large buffer size would require more system memory and would store more packets in buffers with packets waiting in buffers for a longer duration. Choosing smaller buffer sizes causes more packet loss. Therefore, it is necessary to choose a suitable buffer size, which is called the calculated data buffer size (CDBS). The CDBS varies with AHC and N values. Since the packets of the last AHC frames will also stay in the buffers of the nodes in the network, the CDBS is calculated by multiplying the total data rate allocated for data packets (SCDRFWF × number of allocated subchannels for data) by AHC + 1 (number of data packet groups generated in the last AHC frames plus 1 currently generated packet group), as follows:

$$\begin{aligned} CDBS &= (SCDRFWF) \times (number\ of\ allocatedSC\ for\ data) \times (AHC+1) \\ &= (DSPF \times MAU \div (8\ bits\ per\ bytes)) \times (NOS - N \times VSPR_{voice} - N \times VSPR_{video}) \\ &\quad \times (AHC+1)\ bytes \end{aligned} \tag{12}$$

Since there may be [(rate of packets generated by each node in a frame) × (N) × (AHC + 1)] packets in transmission during each frame, this number of slots is needed in the buffer. As long as the number of hop counts is smaller than or equal to the AHC of the system, there will be no packet loss for the calculated number of buffer slots.

The calculated voice and video buffer sizes are formulized as:

$$\begin{aligned} &Calculated_{video/voice}buffer\ size = \left(packet\ size_{video/voice}\right) \times (VPSR_{video/voice}) \times (N) \times (AHC+1)bytes \\ &\text{with } (VPSR_{video/voice}) \times (N) \times (AHC+1)packet\ slots \end{aligned} \quad . \tag{13}$$

3.5. Algorithms for maximizing the network throughput

The proposed MCAF method, which is a combination of time division multiple access (TDMA) and orthogonal frequency division multiple access (OFDMA) methods [12-14], also uses the route reconstruction algorithm, adaptive data rate method, BM method, and SA method. These methods aim to provide fair and cooperative spectrum sharing with BM and minimum hop count for improved throughput per user.

3.5.1. Route reconstruction algorithm

Since the nodes in the network use the random waypoint mobility model [15] with random velocities from 25 km/h up to 40 km/h, it is difficult for the source to predict the complete routes that packets will follow. Therefore, before each packet is forwarded, its route is updated at each node according to the route reconstruction algorithm.

3.5.2. Adaptive rate method

Using the adaptive rate (AR) method, the transmission rate is decreased by the system when congestion and packet losses occur in the network. The maximum data rate per user (MDRPU) in a frame when AR is used can be evaluated by:

$$MDRPU = \left[MSU(bytes/s) \times \frac{TSCFDT}{NOS}\right] \times \left[\frac{1}{N}\right] \times [FL] \times \left[\frac{Succes\ fully\ sent\ packets}{Succes\ fully\ sent\ p. + lost\ p.}\right] bps \quad (14)$$

where MSU is multiplied by the subchannel usage rate of data packets (TSCFDT / NOS) for each node (1/N) in every frame (1/FPS = FL) and by the successful packet transmission rate.

3.5.3. Proposed buffer management method

Once a node starts to send packets, it arranges its buffer such that the packets traveling to the same node are grouped to be sent together and the packet group whose next node has more free memory will be sent first. This process continues during the current OFDMA frame as long as the transmitting node still has packets to send and the frame duration has not expired. In the examples given in Tables 1 and 2, it is assumed that a node can transmit 3 packets in single frame duration.

Table 1. Example of buffer management for node 1 with 6 active nodes.

Stage	Next nodes of the packets	Buffer states of the nodes as percentage of fullness					
1	Node 1 takes spectrum usage turn turn	1	2	3	4	5	6
2	2-3-2-3-5-5-2-2-4-4	44%	28%	31%	42%	34%	14%
3	3-3-5-5-4-4-2-2-2 ->2	42%	30%	31%	42%	34%	14%
4	2-3-3-5-5-4-4-2-2 ->2	40%	32%	31%	42%	34%	14%
5	3-3-5-5-4-4-2-2 ->2	38%	34%	31%	42%	34%	14%

In stage 1 of Table 1, node 1 takes the spectrum usage turn since it has the fullest buffer and needs the spectrum most. In stage 2, the packets of the next node with the emptiest buffer are chosen (packets to node 2) to be sent. In stage 3, a packet is sent to node 2 from node 1. At each transmission, the fullness rates of the buffers are updated (for the example in Table 1, it is increased or decreased by 2%). In stage 4, a new packet, whose next node is node 2, is generated and added to the tail of the queue (note that packet generation of each node is permitted for the rate of total channel capacity / N, such that the packets of all nodes fill the whole spectrum fairly), and a packet is sent from node 1 to node 2. In stage 5, one more packet is sent from node 1 to node 2. Since the current OFDMA frame is over by the sending of 3 packets, the system then chooses the node that will use the spectrum next.

3.5.4. Proposed dynamic spectral aid method

Once a node starts to transmit its packets, if the buffer of the emptiest next node is full, the transmitting node loses its first packet and, regardless of the frame duration state, it returns its spectrum usage rights to the node that needs the spectrum most. At the end of the frame, the spectrum will again be allocated to the node with the most spectral need.

Table 2. Example of dynamic spectral aid for node 1 with 6 active nodes.

Stage	Next nodes of the packets buffer	Buffer states of the nodes as percentage of fullness					
1	Node 1 takes spectrum usage turn	1	2	3	4	5	6
2	2-3-2-3-5-5-2-2-4-4	100%	98%	100%	100%	100%	28%
3	3-3-5-5-4-4-2-2-2 ->2	98%	100%	100%	100%	100%	28%
4	2-3-3-5-5-4-4-2-2 ->2	98%	100%	100%	100%	100%	28%

For the given example in Table 2, in stage 1, spectrum usage is allocated to node 1, which is one of the nodes with a completely full buffer. In stage 2, the packet group for which the next node of packets has the emptiest memory is selected for transmission. In stage 3, the packets to node 2 are placed in front of the queue and one is transmitted. In stage 4, an attempt is made to transmit one packet to node 2; however, since the buffer of node 2 is completely full, the packet is lost and its copy is moved to the tail of the queue. The packet with the emptiest next node is also lost and spectrum usage is now given to the most congested node.

4. Calculation of packet loss rates and throughput

In order to confirm the validity of the simulation results, the throughput and packet loss amounts were also calculated probabilistically for the SA or BM method, or both.

4.1. Calculation of the packet loss rates with N nodes

For MCAF, assuming the total number of packets in the network to be distributed to the nodes proportional to their waiting durations, the packet distribution rates of the nodes can be modeled as $\text{Node}_1 -> 1$, $\text{Node}_2 -> 2\ldots$ $\text{Node}_{N-1} -> \text{N}-1$, $\text{Node}_N -> \text{N}$. Node_1 is considered to be the one that just transmitted its packets and Node_N is considered to be the current transmitter with the fullest buffer. The average packet loss probability (P_{loss}) at one of the remaining N − 1 nodes is calculated as:

$$\begin{aligned} P_{loss} &= \tfrac{1}{N-1} \times P_{lost}(1) + \tfrac{1}{N-1} \times P_{lost}(2) + \ldots + \tfrac{1}{N-1} \times P_{lost}(N-1) \\ P_{loss} &= \tfrac{1}{(N-1)} \left(\sum_{n=1}^{N-1} P_{lost}(n) \right) \end{aligned} \quad , \tag{15}$$

by the sum of probability of sending a packet to each node (probability of 1 / (N − 1)) and the probability of losing the packet at that node (P_{lost} (n)). The term P_{lost} (n) used in Eqs. (15) and (17)-(19) is formulated as:

$$\begin{aligned} P_{lost}(n) &= \frac{\frac{Packet\ distribution\ rate\ of\ node_n}{Sumof\ distribution\ rates\ of\ all\ nodes} \times Total\ packet\ count}{Buffer\ size} \\ &= \frac{\frac{(n)}{\frac{N\times(N+1)}{2}} \times Total\ packet\ count}{Buffer\ size} \\ &= \frac{2\times(n)\times Total\ packet\ count}{Buffer\ size \times N \times (N+1)} \end{aligned} \tag{16}$$

When the proposed BM method is activated (all – SA in Tables 3 and 4), the packet loss rate is calculated as the sum of the probability of having packets to any possible next node combination multiplied by the probability of losing the packet:

$$P_{loss\_BM} = \left[\binom{N-1}{1} p_{lost}^{1} + \binom{N-1}{2} p_{lost}^{2} + \binom{N-1}{3} p_{lost}^{3} + \ldots + \binom{N-1}{N-1} p_{lost}^{N-1}\right] \div \sum_{n=1}^{N-1} \binom{N-1}{n}.$$
$$P_{loss\_BM} = \sum_{n=1}^{N-1} \left[\binom{N-1}{n} \times p_{lost}^{n}\right] \div \sum_{n=1}^{N-1} \binom{N-1}{n} \tag{17}$$

The packet loss rate with the SA method (P_{loss_SA}) (all – BM, in Tables 3 and 4) may be calculated by taking into account the buffer state combination of the remaining N – 1 nodes as the sum of the probability of "choosing a next node with full buffer when only the transmitter has a full buffer (prob. = 0)" plus "when there are 2 nodes with full buffers (including the transmitter) and N – 2 nodes with free buffers" plus... plus "when all buffers are full (prob. = 1)":

$$\begin{aligned} P_{loss\_SA} &= 0 + \left(p_{lost}^{1} \times (1-p_{lost})^{(N-1)-1} \times \tfrac{1}{N-1}\right) + \left(p_{lost}^{2} \times (1-p_{lost})^{(N-1)-2} \times \tfrac{2}{N-1}\right) + \ldots + \\ &\quad \left(p_{lost}^{n} \times (1-p_{lost})^{(N-1)-(n)} \times \tfrac{n}{N-1}\right) \\ P_{loss\_SA} &= \sum_{n=1}^{N-1} p_{lost}^{n} \times (1-p_{lost})^{((N-1)-n)} \times \tfrac{n}{N-1} \end{aligned} \tag{18}$$

When both the BM and SA methods are applied to the system at the same time, we have the following packet loss rate:

$$P_{loss\_rate\_ALL} = \sum_{n=1}^{N-1} \left\{ \left\{ \left[\binom{N-1}{n} \times p_{lost}^{n}\right] \div \sum_{r=1}^{N-1} \binom{N-1}{r} \right\} \times (1-p_{lost})^{((N-1)-n)} \times \frac{n}{N-1} \right\}, \tag{19}$$

by combining Eqs. (17) and (18). The confirmations of packet loss rate calculation and simulation results are given in Figure 3 for N = 6.

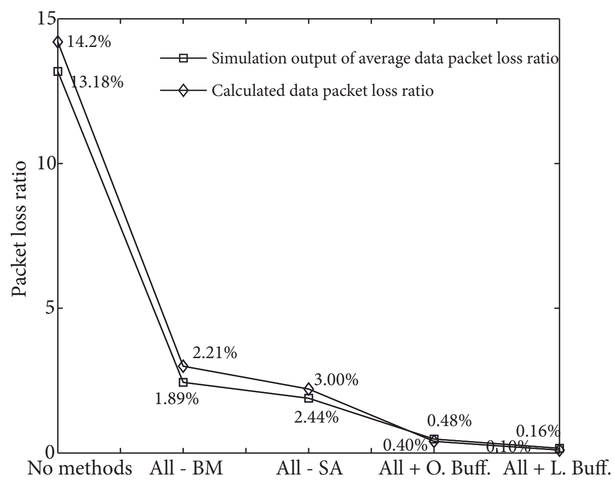

Figure 3. Simulation and calculation results for average data packet loss ratio.

The packet loss ratio in the simulation is determined by:

$$Packet\ loss\ ratio = lost\ packets/(lost\ packets + successfully\ sent\ packets). \tag{20}$$

4.2. Throughput per user calculation with N nodes including bandwidth wastage

The average probability of packet loss at any node in the route is calculated by:

$$p_{OFDMA_{ave}}(N) = \frac{Total\ number\ of\ packets/N}{Buffer\ size} = \frac{Total\ number\ of\ packets}{N \times Buffer\ size}, \tag{21}$$

$$p_{MCAF_{ave}}(N) = \frac{\left(\sum_{n=1}^{N-1} (n) \times \frac{2 \times Total\ number\ of\ packets}{N \times (N+1)}\right)}{(N-1) \times Buffer\ size}, \tag{22}$$

for OFDMA and MCAF, respectively (see Eq. (16)). Since OFDMA uses opportunistic spectrum sharing, the packets in the network will be distributed to N nodes uniformly. For calculation of OFDMA and MCAF generalized throughput values (THR), we need simulation results evaluated for a fixed N value (see Tables 3 and 4 for N = 6) and a set of AHC values for different numbers of nodes evaluated by the routing simulations. Therefore, the throughput per user with bandwidth wastage can be calculated using:

$$THR_{with\_BW\_wastage}(n) = \frac{\frac{AHC_{N_{sim}}\ for\ N\ used\ in\ sim.}{AHC_{n_{sim}}\ for\ n\ nodes} \times [Transmitted\ packets_N\ (bytes)\ includng\ packet\ loss\ rate_N]}{Time_{sim}(s)} Bps$$
$$= \frac{\frac{AHC_{N_{sim}}\ for N_{sim.}}{AHC_{n_{sim}}\ for n nodes} \times \left[\frac{n}{N_{sim}} \times \begin{pmatrix} (sent\ video\ packet_N(bytes)) \\ +(sent\ voice\ packet_N(bytes)) \end{pmatrix} + \begin{pmatrix} ((sent+lost) data\ packet(bytes)_N) \\ \times \frac{TSCFDT_n}{TSCFDT_{N\_sim}} \times (1-P(n)) \end{pmatrix}\right]}{Time_{sim}(s)} Bps, \tag{23}$$

$$\frac{Total\ subchannels\ for\ data\ packets_n(TSCFDT)}{Total\ subchannels\ for\ data\ packets_{SIM}(TSCFDT_{SIM})} = \frac{NOS - (n \times VPSR_{voice}) - (n \times VPSR_{video})}{NOS_{SIM} - (N_{SIM} \times VPSR_{voice_{SIM}}) - (N_{SIM} \times VPSR_{video_{SIM}})}, \tag{24}$$

by taking into account the effects of changes on packet loss rates, average hop count, number of nodes, $VPSR_{voice/video}$, and the number of TSCFDTs. The abbreviation P(n), used in Eq. (23), is used as it is calculated in Eq. (21) or (22) depending on its usage for calculation of THR_{OFDMA} or THR_{MCAF}.

5. Throughput improvement

For each simulation, the improvements in spectrum usage efficiency, data packet loss rate, and throughput of the system were evaluated with and without application of each method for different numbers of nodes. It was shown in [5] that the system throughput increase stabilizes when 4 relay nodes are deployed. Thus, before investigating all of the results evaluated for a range of N values, we focused on results evaluated for N = 6 [7] (including the transmitter and the receiver) as an example. The numeric values of sent/lost packets, spectral usage rates, and throughput values taken from the simulation results for each method are listed in Tables 3 and 4 for N = 6.

When using larger buffer sizes, we expect a positive effect on throughput; however, Table 4 shows that the throughput of this system improved more when CDBS was used. This is due to packets waiting longer in larger buffers without being transmitted. Deactivating the use of the BM method resulted in more congestion and packet loss at buffers of nodes and decreased system throughput performance. The throughput improvement was the worst when the SA method was excluded in simulations (see Table 4). The most important criteria influencing system throughput are effective spectrum usage (given in Table 4), the packet loss ratio (given in Table 3), and AHC. The most effective spectrum usage and greatest improvements to throughput are achieved when all proposed methods are active and CDBS is used (see Table 4).

Table 3. Simulation results of number of sent/lost packets and packet loss ratios

Simulation output data in 5 s (for N = 6)	Average effective spectrum usage (Bps)	Average effective spectrum usage (%)	Average overall throughput (Bps)	Improvement %	Throughput improvement loss by not applying the method (improvement of the method)
No methods	2,642,358	90%	1,225,488	0%	27%
All – AR	2,790,893	95%	1,489,122	22%	5%
All – SA	2,782,438	95%	1,308,980	7%	20%
All – BM	2,751,435	94%	1,412,112	15%	12%
All + larger buffer	2,808,181	96%	1,519,210	24%	3%
All + CDBS	2,801,605	96%	1,556,458	27%	0%

Table 4. Simulation results for different methods including bandwidth wastage.

Simulation output data in 5 s (for N = 6)	Number of video packets		Number of voice packets		Number of data packets		Packet loss ratio	
	Sent	Lost	Sent	Lost	Sent	Lost	Video Voice	Data
No methods	494	0	2389	0	29344	4456	0%	13.18%
All – AR	507	0	2576	0	35953	653	0%	1.78%
All + larger buffer	515	0	2532	0	36691	58	0%	0.16%
All – BM	517	0	2582	0	33978	654	0%	1.89%
All – SA	502	0	2575	0	31419	786	0%	2.44%
All + CDBS	515	0	2544	0	37631	182	0%	0.48%

In the simulation process, it is assumed that each node uses at least 1 separate subchannel for every 4 frames [12] (with $\mathrm{VPSR}_{video} \geq {}^{1}/_{4}$) for its video conversations and at least 1 separate subchannel in every frame [12] (with $\mathrm{VPSR}_{voice} \geq 1$) for its voice conversations. Thus, more than $[\mathrm{NOS} - (\mathrm{N} \times \mathrm{VPSR}_{voice} + \mathrm{N} \times \mathrm{VPSR}_{video})]$ subchannels will be used by N nodes for data packets, and we have $\{\mathrm{NOS} - [\mathrm{N} \times 1 + (\mathrm{N} \times {}^{1}/_{4})]\} \geq 0$, $\mathrm{N} \leq 24$ for NOS $= 30$. Therefore, N is increased up to 24 in the simulation.

The simulation results of throughput for different values of N are given in Figure 4. Furthermore, these results were evaluated for corresponding AHC and video/voice packet sending rates ($\mathrm{VPSR}_{video}/\mathrm{VPSR}_{voice}$) at that instant of the simulation. Figure 4 shows that our pure system simulation results are confirmed by results evaluated in the literature and results from the calculations that we carried out. Figure 4 also shows that simulation results of pure MCAF without AR, BM, and SA exactly match the calculation results of pure

MCAF without AR, BM, and SA. The calculation results of pure OFDMA without AR, BM, and SA exactly match the unicast analysis results of pure OFDMA without AR, BM, and SA [9]. It is shown in Figure 4 that the proposed BM and SA methods, used with MCAF, improve the system throughput performance by up to 36% for N = 4 when all methods are active and CDBS is used. The results of [10] in Figure 1b, which exactly match the results of [9] and [11], are continued in Figure 4 with the legend "Unicast analysis results of pure OFDMA without AR, BM, and SA" to show that the results of [9-11] match the results evaluated in this study for the pure system.

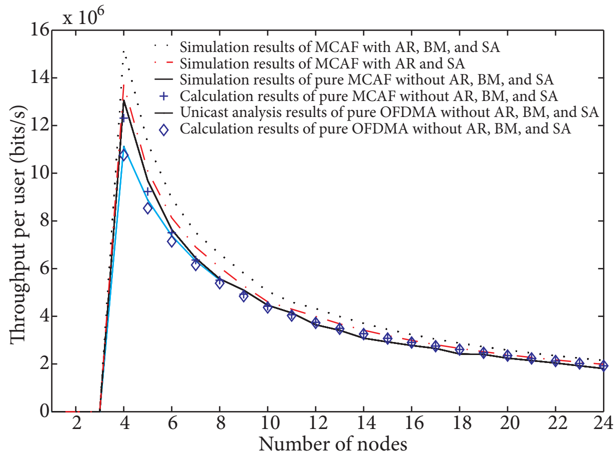

Figure 4. Throughput results of OFDMA, MCAF, and unicast asymptotic analysis.

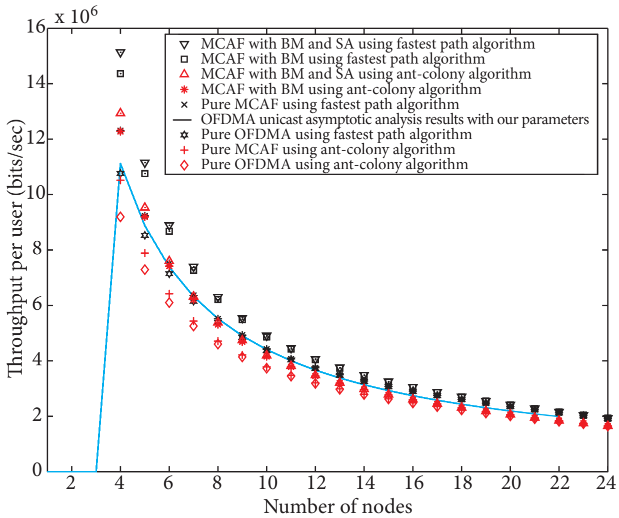

Figure 5. The throughput performances of routing algorithms.

6. Effects of routing algorithms on system throughput

The routing algorithm used in the simulation and the AHC value it provides has great importance because of its effect on the resultant throughput (see Eqs. (6) and (23)). Therefore, routing simulation programs implementing the fastest path [16] and ant colony [17] routing algorithms were developed in MATLAB. Both routing algorithms are embedded in our system, running simultaneously, and they make their own decisions under the same conditions.

The proposed methods improve the throughput when used with either the ant colony or the fastest path routing algorithm. Since the fastest path routing algorithm generates routes with smaller AHC values, it always achieves greater output, even for the pure system (see Figure 5). The fastest path routing algorithm is therefore preferable for our proposed methods.

7. Conclusion

An event-driven simulation study was presented for designing a relay-based WiMAX system that directly focuses on decreasing packet loss ratio and improving the throughput of a cognitive multimedia network. The proposed MCAF scheme is a combination of the OFDMA and TDMA methods. It uses AR transmission and route updating methods in addition to the proposed SA and BM methods. The simulation results for the throughput of a conventional 802.16j network were confirmed. Packet sizes and buffer sizes were calculated and formulized for different data types. By adjusting the packet size, the bandwidth wastage caused by unfilled subchannels was minimized.

First, it was shown that optimizing the buffer size provides better throughput performance than the throughput performance evaluated when the buffer size was doubled. By use of the MCAF method (with AR, BM, and SA), which does not require major structural modifications on the existing system, the packet loss ratio was decreased, the packet generation rate was increased, and the throughput was improved.

Second, the probabilistic calculation of packet loss rates and throughput values were presented. It was shown that the packet loss rate and the throughput calculation results matched the simulation results.

Finally, throughput was improved for both the ant colony and fastest path routing algorithms, with the fastest path algorithm achieving greater outputs.

The results of this work may also be evaluated using other event-driven network simulators in the area and may be generalized for other protocols used for wireless mobile and multihop relaying networks.

8. Discussion and future work

In this work, the throughput of a mobile ad hoc network was improved by use of novel cognitive methods. It was shown that the proposed methods achieved throughput improvement in the systems either by using ant colony or fastest path routing algorithms. In the future, the effects of long-life routing algorithms, such as associativity-based routing or enhanced associativity-based routing, on each proposed throughput improved method can be investigated for different vehicular speeds.

All of the methods, formulations, and results of this study can also be used in designing or analyzing a unicast mobile multimedia network.

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