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Flexible and efficient aggregation framework for antifragile wireless mesh networks

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Abstract Wireless mesh networks (WMNs) have found an interesting alternative application in public safety and disaster recovery as they are enabled with key features such as fault tolerance, broadband support, and interoperability. However, their sparsely distributed wireless nodes need to frequently share the control packets among each other for successful data transfer. These packets give rise to a considerable amount of control overhead, especially for multimedia traffic, which is not bearable in jeopardy situations of network disaster such as 9/11 and Hurricane Katrina. Hence, to avoid the huge cost of control overhead, aggregation is supposed to be one of the handy solutions for building a new object from one or more existing objects of network traffic. Generally, aggregation has three types to be executed on 'packets', 'frames' and the 'links' of a network. The network decision that when and which type of aggregation is suitable in a given scenario is a complex balancing act. Because, it is bounded by various live statistics of the communication link (e.g., buffer size, link quality, bandwidth, maximum transmission unit, and delay etc.) and thus, can considerably affect the network efficiency. If such statistics do not support aggregation or its certain type, network performance may cause worse affects. This paper proposes a decision-oriented dynamic solution, namely AADM, which is an adaptive aggregation-based decision model. Based on the link quality and its live statistics, AADM dynamically takes judicious decisions about aggregation and its specific type to achieve the desired outcome. Despite network scalability, quality of service, power optimization, and network efficiency, it also reduces control traffic in WMNs. OMNET++ simulations are used to verify AADM. Simulation results have shown that AADM outperforms existing static approaches in terms of packet loss, throughput and delay.

Keywords Disaster recovery · Packet aggregation · Adaptive aggregation · WMN · MTU and VoIP

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1 Introduction

Disasters such as earthquakes and floods affect large geographical areas, which creates fragile environment for rescue services to operate. One of the major aftermaths of these disaster situations is lack of communication infrastructure to support applications that could provide coordination among all the rescue teams and disseminate necessary information and data for informed decision making. This requires an antifragile and resilience application framework that can also provide on-demand computing resources and judicious decision making. Given this opportunity, wireless mesh networks

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(WMNs) have found an interesting alternative application in public safety and disaster recovery (PSDR) as they are enabled with key features such as fault tolerance, broadband support, and interoperability.

Typically a WMN is made up of radio nodes organized in a mesh topology. Whereby, a WMN is a group of unplanned nodes and mesh routers interconnected through wireless links [1]. The wireless service access point is installed at each network user's locale requiring minimal configuration and infrastructure. In WMN, each network acts as an autonomous user and forwards data traffic to the next node. The network infrastructure is decentralized, in which these autonomous nodes send and receive data independently and is not controlled by the centralized controller.

A WMN usually consists of three types of nodes: mesh clients (MC), mesh router (MR), and mesh gateway (MG). Clients can be either stationary or mobile, and can form a mesh network among them by involving the mesh routers. Whereas the wireless routers in WMN relay packets originating from the client nodes to the gateway nodes, which are further connected to a backbone wired network. In this way, large areas can be covered by a low-cost infrastructure. As an intermediary, a mesh access points (MAP) also serves the network, which works on the principle of a radio frequency (RF) access point and transmits the packet to other mesh networks, or to the wired backbone. It takes the packet form one mesh client network and forwards either to some other mesh client network or some heterogeneous network using the wired backbone. MR chooses a suitable gateway to route the received packets. MG is an entity that combines or integrates two networks like connecting the wired network with the wireless mesh network for better support.

Wireless mesh networks also inherit the characteristics of ad hoc networks such as self-organized, self-configured, self-healing, scalability and reliability. Such kinds of networks automatically incorporate a new node into the existing structure without requiring any adjustments by a network administrator. Its configurations allow a local network to run faster, because a local packet does not need to travel back to a central server. They can provide cost-effective network deployment under the diverse environments of PSDR and better coverage to both stationary and mobile users. They offer multiple paths from a source to destination through their on demand routing algorithms allowing each node to make an intelligent decision i.e., which path can effectively be used to forward packets through the network in order to improve the overall network performance.

While WMN's deployments for PSDR purpose, efficient utilization of the bandwidth is a critical issue especially for audio, text, images or video traffic. Some types of data might be bearing small chunk of packets for control information, though small but numerous packets give rise to

considerable amount of overhead for control information in wireless networks [2]. Moreover, such increased overhead is also not affordable in jeopardy situations such as 9/11 and Hurricane Katrina. This paper addresses the state-of-the-art aggregation approaches bringing about some imminent modifications that would not only increase throughput but could also use to produce the optimum results in regard to network performance.

We proposed a flexible and efficient aggregation framework for WMN named as adaptive aggregation based decision model (AADM) and is supported by OMNeT++ simulation. In AADM the aggregation is performed, based on run time decisions after evaluating the more dynamic factors like congestion, routing, reliability, average delay and energy consumption. The AADM exclusively monitors the live statistics of a network channel and takes run time decisions that either aggregation is needed or not? If needed, whether it needs to aggregate on real time or non-real time basis? Then it defines the maximum transmission unit (MTU) for aggregation depending on aforementioned factors. Finally, it decides which type of aggregation is suitable for the aggregation i.e., node to node or end to end aggregation. AADM also defines role of source and destination in the aggregation applied.

The rest of this paper is organized in the following order: Related work is elaborated with respect to types of aggregation approaches in Sect. 2. Proposed framework has been described containing the detail of aggregation algorithm for generating hash tables in Sect. 3, and then results and analysis follow in Sect. 4, and in the final section we conclude the paper.

2 Related work

Aggregation is a composition technique for building a new object from one or more existing objects that support some or all of the new object's required interfaces [2]. There are two types of aggregation with respect to packet size. One is static aggregation and the other is dynamic aggregation. In static aggregation, the maximum transmission unit (MTU) remains fixed, while in dynamic aggregation it may varies according to the input parameters. The dynamic aggregation is similar to the fixed aggregation except the characteristics of local link it used to determine an appropriate packet size to reduce the chances of network packets being dropped. Such packet size is called the aggregation threshold and is not more than that the value of MTU. The decision when to aggregate is substantially influenced by two parameters i.e., the maximum queue size and the time delay. In this regard, existing literature can be broadly classified into three categories i.e., frame aggregation, packet aggregation and the link aggregation.

2.1 Packet aggregation

In packet aggregation a number of small packets are combined into larger packets. In this approach distributed aggregators are used to collect smaller packets through various network connections and assemble as a large packet. In packet aggregation the sender adds an aggregation header, so that the receiver can de-aggregate the packets correctly, which cause this a more critical operation. There are two types of packet aggregation i.e., hop-by-hop aggregation and end-to-end aggregation. In hop-by-hop scheme, the packets are aggregated and de-aggregated on each hop frequently until packets become arrived at the final destination; while in end-to-end scheme packets are aggregated at source and de-aggregated at destination node only.

Bayer et al. [2] proposed a hop-by-hop packet aggregation mechanism for 802.11s WMNs and worked on the feasibility of voice over internet protocol (VoIP) in a dual radio mesh environment. They introduced a novel packet aggregation scheme to avoid the small packet overhead that reduces the MAC layer busy time and enhances the network performance. It aggregates packets at IP layer and is enabled with different network conditions and traffic characteristics. This approach does not increase delay unless it provides a good aggregation ratio.

Kyungtae et al. [3] developed a distributed multi-hop aggregation algorithm for VoMESH that uses natural waiting time in the interface queue of packets in a loaded network. They used a combined approach of header compression and packet aggregation that reduces VoIP protocol overhead and introduces signalling overhead. They proposed a zero-length header compression algorithm integrated with packet aggregation, which does not need to depend on the signalling mechanism to recover the context discrepancy between compressor and de-compressor. The major challenges are to reduce VoIP protocol overhead and 802.11 MAC overhead, which decreases VoIP performance for mesh network. They investigate these problems in 802.11-based wireless mesh network and propose novel solutions to reduce overheads. Each of these methods produces considerable improvement in operation of the mesh with respect to network capacity and quality of service (QoS).

Andreas et al. [4] proposed an adaptive hop-by-hop aggregation scheme that computes the target aggregation size for each hop and is based on wireless link characteristics. This scheme behaves well relative to its counterpart schemes, like static aggregation scheme for various performance parameters. They focused on the relation between link quality and packet size for packet aggregation in multi-hop WMNs. Therefore, the overall aggregation along a path will not be constrained by the weakest link which leads to significant performance improvement. Marcel et al. [5] proposed a novel packet aggregation mechanism which leads to the enhancement of VoIP capacity with the maintenance of voice quality. It helps reduce the MAC layer contention and significantly increases the concurrent VoIP flows. It is a hop-by-hop aggregation scheme and inherently derives all the limitations of such schemes. Their main contribution is to design the packet aggregation mechanism which adapts itself to network traffic and minimizes delay and overhead without requiring any changes to current MAC layer implementations. So then, increases VoIP performance and reduces MAC delay.

The above solution is focused on increasing the VoIP traffic in multi-hop WMNs. Niculescu et al. [6] proposed several methods to improve voice quality and used multiple interfaces label based forwarding architecture along with path diversity and aggregation. They present experimental results from an IEEE 802.11b test bed, optimized for voice delivery. They implemented a distributed packet aggregation strategy and made a good use of the natural waiting time of MAC for aggregation purpose. These methods show a little improvement in the enhancement of simultaneous number of calls. They focus on two important problems in supporting VoIP over WMNs i.e., increasing VoIP capacity and maintaining QoS under internal and external interference. They evaluate the performance of VoIP, over WMN and provide various approaches for system optimization.

Raghavendra et al. [7] proposed another method on an IP-based adaptive packet concatenation for WMNs. In this method authors used the adaptive technique i.e., to decide during runtime whether to concatenate the packets or not. The packet size for aggregated packets is calculated based on the route quality, because a good quality route can potentially carry larger aggregated packets.

In our findings, the aforementioned solutions have following shortcomings: [2,3] and [6] solutions are static and are inefficient to use if aggregation is not required. The [2] and [5] solutions ignore the bandwidth, energy consumption, congestion, routing and other important factors in packet aggregation. The [3,5] and [6] solutions cannot be applied in single hop WMNs. Since, [5] ignores the security of data and the network at the same time. The [5] and [6] emphasize on increasing capacity and QoS in VoIP, however, later in particular, did not specify routing mechanism of the packets. The [7] ignores the influence of link quality on packet size or only consider the end-to-end path quality due to the use of routing metrics such as WCETT. These metrics reflect path characteristics, suitable for end-to-end aggregation and thus achieve suboptimal performance. If there is a bottleneck link such as one characterized through low signal quality, this will lead to a small packet size for end-to-end aggregation, and thus the benefit of aggregating will be lost or negligible at all.

Furthermore, the solutions proposed for VoIP cannot successfully be implemented in WMN due to the absence of a central controller. Secondly, the main purpose of these solu-

tions is to increase the number of calls, improve voice quality and minimize overhead. Moreover, many other important parameters such as security, congestion, and routing, reliability and energy consumption and delay are ignored. Therefore, these solutions are not suitable for aggregation in WMNs. Some solutions are particularly suggested for wireless sensor networks (WSNs) [8–12]. They are not applicable in WMNs because of manifold reasons. Firstly, these solutions are proposed for WSNs, which cannot be implemented in WMNs because WSNs has a central controller while WMNs are essentially lacking any central device. Secondly, the solutions proposed for WSNs mainly focused on energy consumption. Thirdly, all the above solutions are static and cannot decide on runtime whether aggregation is required or not in a given scenario, which make these solutions inefficient especially for WMNs.

2.2 Frame aggregation

Frame aggregation integrates many smaller packets together to form a single large packet that could be sent in a single transmission and avoids the header of individual packets. The frame aggregation scheme helps in improving the efficiency for Media Access Control (MAC) protocol. Lee et al. [13] proposed a hop-by-hop frame aggregation scheme for VoIP that reduces interference effect by combining the inter-call aggregation and the pseudo-broadcast parameters. This scheme works effectively not just for the calls traveling same routed path but also for calls crossing each other inside the multi-hop networks. In this way the calls could be entertained not only for the same routing path but also for the paths crossing with other routing paths inside the multi-hop networks.

Lin et al. [14] proposes a dynamic frame aggregation scheme called dynamic aggregation selection and scheduling algorithm. This algorithm dynamically adopts a suitable scheme out of different aggregation mechanisms and is based on waiting for the next frame for the aggregation method. Kim et al. [15] proposed another solution for frame aggregation, which improves the throughput performance of IEEE 802.11 WLAN. They devised a simple method to implement the frame aggregation into the real test-bed using off-the-shelf IEEE 802.11 wireless LAN products via device driver modifications. This scheme aggregates multiple frames into a single MAC frame in order to reduce MAC/PHY overheads of the IEEE 802.11 WLAN. Their current frame aggregation scheme works when frames are transmitted to the same receiver.

Ganguly et al. [16] proposed a frame aggregation scheme for IEEE 802.11 WLANs in which, a model is proposed for calculating successful transmission probability of the frames with certain length. Simulation results shown that network contention level have minor influence over the transmis-

sion. The aggregation scheduler scheme by [17], presents a detailed simulation study. The proposed frame aggregation based scheduler dynamically chooses the aggregated frame size and technique, based on various relevant parameters. The frame aggregation techniques have to be effective even under unsaturated network conditions with limited throughput. This technique estimates the time deadline for frame transmission. In addition, it also outlines a method for selecting the frame aggregation type. Lin et al. [18] conduct a thorough study of Aggregated MAC Service Data Unit (A-MSDU) and Aggregate-MAC Protocol Data unit (A-MPDU) frame aggregation schemes in IEEE 802.11n. They propose a simple optimal frame size adaptation algorithm for A-MSDU and A-MPDU under error-prone channels.

Another adaptive approach is proposed by [19]. In which, authors used a cross-layer scheme that improves scalability of WMNs using aggregation of MAC layer frames. The aggregation is performed on top of MAC layer, allowing us to reduce the overhead caused by protocol headers and the contention mechanism regulating the IEEE 802.11 standard. This approach relies on adaptive aggregation scheme and leverages the channel probing functionalities of mesh routers. Such information is exploited in order to compute the optimal saturation burst length.

Some adaptive approaches are also proposed by [7] and [19]. These solutions are adaptive in nature but just took very limited parameters in consideration. Also these solutions did not decide which type of aggregation is suitable for the given scenario. Some authors in [20] and [21] have proposed mechanisms to solve the performance anomaly problem by packet aggregation using a dynamic time interval, which ultimately depends on the busy time of the wireless medium.

The shortcomings of above schemes, in our opinion, are given as under: Firstly, a hop-by-hop frame aggregation (HHA) is proposed for VoIP in [13]. Secondly, the solution is focused on increasing in number of calls, bandwidth utilization, routing mechanism, energy consumption and other important parameters are ignored. Thirdly, hop-by-hop aggregation solution is not efficient as aggregation and de-aggregation is performed at every hop, which utilized the network resources relatively higher.

Almost all of these frame aggregation schemes except [14] and [19] are misfit for WMNs due to fundamental difference in their network architecture. Furthermore, all of these schemes except [13] are static, which increases inefficiency, consumes energy and bandwidth competitively more when aggregation is not required, and thus reduces the network life time due to higher energy consumption. At the same time these schemes cannot decide whether aggregation is required or not, which makes these solutions inefficient for a dynamic environment. In [14,15,15,17–19] schemes, the aggregated frames will be lost in case of network congestion as there is no acknowledgment in data link layer like of wire-

less networks. In addition, [15] relies on a centralized access point for being a centralized scheme and can be taken as a flaw; likewise, [20] and [21] proposed solution for a complete decentralized infrastructure.

While [17] scheme reduces the efficiency of MAC layer in a noisy environment. At the same time it ignores energy consumption, congestion and reliability. In [14,15,15,17–19] the data link layer based solution is not secure enough. In [19] the aggregation of frames increases the transmission rate of data. Where, if the frame size is greater than MTU, data will ultimately be lost, as there is no option of fragmentation in data link layer is available. Similar case holds for [14,15,15,17], and thus decrease the overall efficiency of the network. In [20] and [21], the authors used average count, and sum for aggregation in dynamic environment, but they simultaneously ignores other significant constraints like energy consumption, congestion and routing.

2.3 Link-based aggregation

Link aggregation aggregates many single links together to form a link aggregation group, so that the whole link group would be treated as single link by any MAC client. In this way the logical links from node to node may be established consisting more than one parallel instance of full duplex point-to-point links. In this way the increased capacity of independent link aggregation from a MAC layer is likely to achieve.

Okech et al. [22] proposed a link-based aggregation framework and compares its performance with the fixed aggregation and no-aggregation based schemes. They computes packet size by considering the link-based characteristics and analyzed that VoIP performance can considerably be enhanced by adjusting packet size with link characteristics.

3 Proposed scheme

In this paper, we proposed a flexible and efficient aggregation framework for WMNs, called adaptive aggregation-based decision model (AADM). In which, aggregation is based on various parameters i.e., congestion, delay, packet loss ratio (PLR), bit error rate (BER), buffer size and bandwidth. It dynamically considers that aggregation is needed or not, if needed and then it will decide which type of aggregation is suitable, i.e., node-to-node or end-to-end aggregation. Later, it decides whether system could aggregate both real time and non-real time data subject to the bandwidth support; otherwise, it prefers real time traffic only. Finally, AADM specifies MTU for aggregation by investigating live statistics of data traffic. Moreover, it defines the role of source and destination in the aggregation mechanism applied.
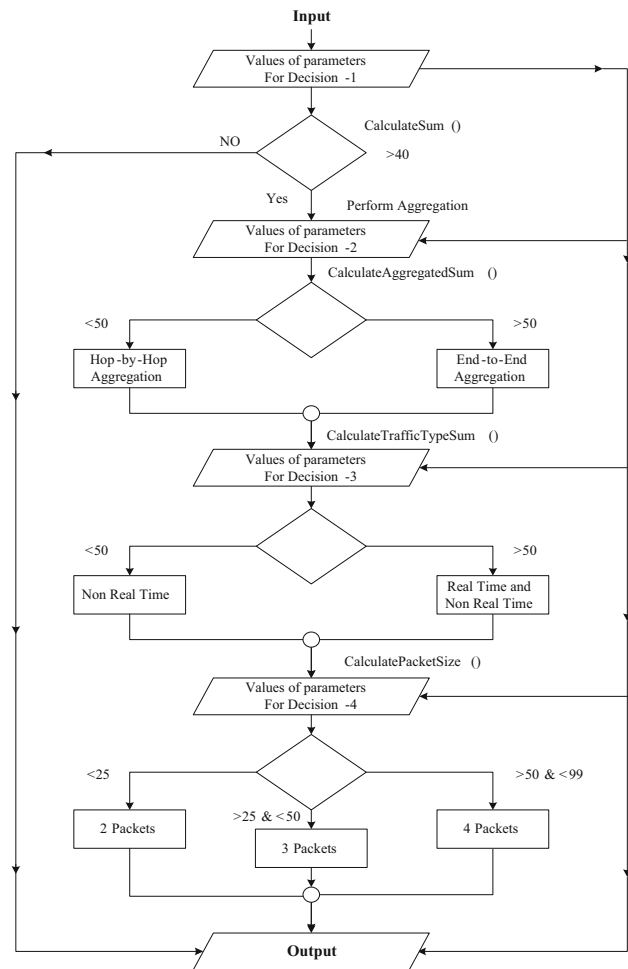

Fig. 1 Flow graph for AADM

Figure 1 describes the flow graph of proposed algorithm and the necessary steps that generate aggregated packet. In which, it described that AADM first calculates the parameters sum to find the chances of aggregation being performed or not. In case the previous condition becomes true, it further calculates the sum of parameters again to select the type of aggregation, and finally it calculates size of aggregated packets to be delivered to the destination. There are countless factors that can be taken into account with reference to aggregation. However, we emphasize a few of the primary factors like congestion, delay, packet loss ratio, bit error rate, buffer size and bandwidth. Here, we assume that all of the remaining factors are contributing average and acceptable values, and are sufficient enough to make the aggregation feasible. Hence, keeping other entire factors constant with respect to source nodes physical context, we shall focus on the above mentioned primary factors. The impact of these primary factors is sufficient enough to contribute positively towards the aggregation.

In AADM, we developed an intelligent algorithm called aggregation finding possibility algorithm (AFPA) as demon-

strated bellow in Algorithm 1, which exclusively identify possibility of aggregation based on the information provided by an environment. If aggregation is not feasible then it recommends transferring the packets as it is, otherwise AFPA has to take few more decisions.

In this connection, AFPA tells whether it should be hop-by-hop aggregation or it should be end-to-end aggregation, packets of which source and destination should be aggregated and so on. These decisions make the aggregation more efficient and results in delivery of more and more packets by consuming minimum number of resources. This solution also helps in load balancing over the network and indirectly plays its role to reduce the network congestion.

The packet aggregation in AADM decreases physical and MAC layer overhead and thereby reduces the transmission time. At times the aggregation might not be required and will be decided by AADM. The proposed algorithm takes some of parameters as the input and gets helps to decide with a threshold value, say, THRESH_LIMIT_AGG, which is set as 40 in our scenario (assumed, and can be adjusted according to the specification of network). If the sum of the parameters is greater than the THRESH_LIMIT_AGG, the aggregation will be performed. If the value of THRESH_LIMIT_AGG is less than 40 the original packets will be sent without the aggregation process. It takes the input parameters like congestion, bit error rate, delay, bandwidth, and buffer size and packet loss ratio. If there is high congestion on the channel, the algorithm adaptively discourages the aggregation and vice-versa. The higher round trip time (RTT) or the increased degree of packet drop due to the short queue buffer indicates the higher degree of congestion over the path. At the same time, if there is more routing delay which is due to the heavy routing tables, no aggregation will be performed.

If there is high average delay due to less buffer or queue size, the aggregation is disallowed by the algorithm. Furthermore, if the link is not reliable with respect to bit error rate, the aggregation will not be performed, as the link with high bit error rate cannot be relied upon and may negatively affect about integrity of data. Hence, the link quality (i.e., in relation to BER) is considered another significant factor in deciding the viability of aggregation.

Lastly, the number of the neighbours might affect such decision given that, the increased number of neighbours performing aggregation would be a bottleneck for the bandwidth capacity. If the bandwidth is reasonably higher, the aggregation probability would be relatively high. Hence, the higher number of neighbours does not negatively affect the system. The higher the packet loss ratio, the lesser would be the chances of aggregation, which is one of the higher weight parameters that can aggressively affect the chances for aggregation. For all the decisions, there is a reasonably higher weight scale for input parameters, that is, varying values will be fed to the algorithm.

In turn, it gives the sum of parameters for comparison with THRESH_LIMIT_AGG value. This threshold value will help in deciding the possibility of performing aggregation.

Adaptive aggregation based decision model considers another threshold value namely THRESH_LIMIT_TYPE that is set to 50 %, which suggests the type of aggregation to be performed. The type of aggregation is selected from hop-by-hop and end-to-end aggregation. The high sum of input parameters value suggests the end-to-end type of aggregation; otherwise hop-by-hop aggregation is ultimately considered to execute. Hence all these threshold values allow AADM while making such critical decisions.

The varying degree of input values may lead in the attainment of THRESH_LIMIT_AGG up to 40 % and progress to further decisions required. In one case less network congestion might be contributor in the achievement of such threshold. In other case, if there is high congestion, some other parameters like less average delay, higher buffer size, less packet drop ratio, higher bandwidth or less bit error rate, might be the major contributors in achieving THRESH_LIMIT_AGG and vice-versa. The same threshold is used to measure the possibility of real and non-real time traffic aggregation. The real time traffic would be aggregated in every case while non-real time traffic depends on the reasonable value of threshold. AADM considers another threshold called PACKET_SIZE_THRESHOLD (PST) i.e., PST_1, PST_2 and PST_3. This threshold estimates the range of maximum packet size, in which the sum of the parameters exists. The larger sum would certainly lead to higher threshold, and thus the higher number of packets per aggregation.

Adaptive aggregation based decision model is fundamentally based on fuzzy logic, as the real values of the parameters are passed as input arguments to the algorithm it transforms all these inputs into a suitable value for making decisions. Let us call it as the F_L_value for applying fuzzy logic. These values correspond with the actual or real parameters. These F_L_values are further utilized to calculate the ultimate threshold value. There are different levels of F_L_values assigned to different parameters in every decision. In most cases the relationship is inversely proportional between the original input value and the assigned weight scale.

For instance, in the first case of decision if congestion is very low i.e., 1 or 5 % we assign 0.15–0.20 points and if congestion is very high i.e., 20–25 % then assign 0.01–0.02 points etc. Similarly, for each parameter F_L_value has been assigned. Here, the total points are 20 out of which the F_L_values are given. Likewise, in packet loss ratio and bit error rate the total points are 25, in delay 10, in bandwidth 10 and buffer size is also 10. These maximum points vary for every decision and weight formula also changes for every decision.

Algorithm 1 AFPA: Aggregation finding possibility

1:	AFPA (PLR, BER, D, B, BS, Cn){
2:	If PLR < (1%) or = (1%) then p = 0.25
3:	ElseIf PLR > (1%) and < or = (2%) then p = 0.22
4:	ElseIf PLR > (2%) and < or = (5%) then p = 0.15
5:	ElseIf PLR > (5%) and < or = (10%) then p = 0.10
6:	ElseIf PLR > (10%) and < or = (15%) then p = 0.07
7:	ElseIf PLR > (15%) and < or = (20%) then p = 0.02
8:	ElseIf PLR > (20%) and < or = (25%) then p = 0.01
9:	If BER < (1%) or = (1%) then Lq = 0.25
10:	ElseIf BER > (1%) and < or = (2%) then Lq = 0.22
11:	ElseIf BER > (2%) and < or = (5%) then Lq = 0.15
12:	ElseIf BER > (5%) and < or = (10%) then Lq = 0.07
13:	ElseIf BER > (10%) and < or = (20%) then Lq = 0.02
14:	ElseIf BER > (20%) and < or = (25%) then Lq = 0.01
15:	If D < 10ms or = 10ms then d = 0.1
16:	ElseIf D > 10ms and < or = 100ms then d = 0.07
17:	ElseIf D > 100ms and < or = 500ms then d = 0.04
18:	ElseIf D > 500ms and < or = 1s then d = 0.02
19:	ElseIf D > 1s and < or = 2s then d = 0.01
20:	If B < 512Kb or = 512Kb then b = 0.1
21:	ElseIf B > 512Kb and < or = 1Mb then b = 0.09
22:	ElseIf B > 1Mb and < or = 2Mb then b = 0.07
23:	ElseIf B > 2Mb and < or = 4Mb then b = 0.04
24:	ElseIf B > 4Mb and < or = 8Mb then b = 0.02
25:	ElseIf B > 8Mb and < or = 16Mb then b = 0.01
26:	If BS < 100 or = 100 then q = 0.1
27:	ElseIf BS > 100 and < or = 500 then q = 0.07
28:	ElseIf BS > 500 and < or = 1000 then q = 0.04
29:	ElseIf BS > 1000 and < or = 5000 then q = 0.01
30:	If Cn < (1%) or = (1%) then c = 0.2
31:	ElseIf Cn > (1%) and < or = (5%) then c = 0.15
32:	ElseIf Cn > (5%) and < or = (10%) then c = 0.10
33:	ElseIf Cn > (10%) and < or =(15%) then c = 0.06
34:	ElseIf Cn > (15%) and < or = (20%) then c = 0.02
35:	ElseIf Cn > (20%) and < or = (25%) then c = 0.01
36:	SAF = ∑(b, p, d, Lq, q, c)
37:	If SAF < THRESH_LIMIT_AGG
38:	Then "Aggregation cannot be performed"
39:	Else {{
40:	If PLR < (1%) or = (1%) then p = 0.35
41:	ElseIf PLR > (1%) and < or = (2%) then p = 0.30
42:	ElseIf PLR > (2%) and < or = (5%) then p = 0.20
43:	ElseIf PLR > (5%) and < or = (10%) then p = 0.10
44:	ElseIf PLR > (10%) and < or = (15%) then p = 0.07
45:	ElseIf PLR > (15%) and < or = (20%) then p = 0.02
46:	ElseIf PLR > (20%) and < or = (25%) then p = 0.01
47:	If BER < (1%) or = (1%) then Lq = 0.35
48:	ElseIf BER > (1%) and < or = (2%) then Lq = 0.30
49:	ElseIf BER > (2%) and < or = (5%) then Lq = 0.20
50:	ElseIf BER > (5%) and < or = (10%) then Lq = 0.10
51:	ElseIf BER > (10%) and < or = (15%) then Lq = 0.07
52:	ElseIf BER > (15%) and < or = (20%) then Lq = 0.04
53:	ElseIf BER > (20%) and < or = (25%) then Lq = 0.01
54:	If D < 10ms or = 10ms then d = 0.05
55:	ElseIf D > 10ms and < or = 100ms then d = 0.04
56:	ElseIf D > 100ms and < or = 500ms then d = 0.03
57:	ElseIf D > 500ms and < or = 1s then d = 0.02
58:	ElseIf D > 1s and < or = 2s then d = 0.01
59:	If B < 512Kb or = 512Kb then b = 0.05
60:	ElseIf B > 512Kb and < or = 1Mb then b = 0.05
61:	ElseIf B > 1Mb and < or = 2Mb then b = 0.04
62:	ElseIf B > 2Mb and < or = 4Mb then b = 0.03
63:	ElseIf B > 4Mb and < or = 8Mb then b = 0.02
64:	ElseIf B > 8Mb and < or = 16Mb then b = 0.01
65:	If BS < 100 or = 100 then q = 0.1
66:	ElseIf BS > 100 and < or = 500 then q = 0.07
67:	ElseIf BS > 500 and < or = 1000 then q = 0.04
68:	ElseIf BS > 1000 and < or = 5000 then q = 0.01
69:	If Cn < (1%) or = (1%) then c = 0.10
70:	ElseIf Cn > (1%) and < or = (5%) then c = 0.07
71:	ElseIf Cn > (5%) and < or = (10%) then c = 0.05
72:	ElseIf Cn > (10%) and < or = (15%) then c = 0.03
73:	ElseIf Cn > (15%) and < or = (20%) then c = 0.029
74:	ElseIf Cn > (20%) and < or = (25%) then c = 0.01
75:	SHE = ∑(b, p, d, Lq, q, c)
76:	IF SHE > THRESH_LIMIT_TYPE
77:	Then "End-end-based approach is selected"
78:	Else "Hop-based approach is selected"} {
79:	If PLR < (1%) or = (1%) then p = 0.20
80:	ElseIf PLR > (1%) and < or = (2%) then p = 0.17
81:	ElseIf PLR > (2%) and < or = (5%) then p = 0.12

82:	ElseIf PLR > (5%) and < or = (10%) then p = 0.10
83:	ElseIf PLR > (10%) and < or = (15%) then p = 0.07
84:	ElseIf PLR > (15%) and < or = (20%) then p = 0.02
85:	ElseIf PLR > (20%) and < or = (25%) then p = 0.01
86:	If BER < (1%) or = (1%) then Lq = 0.20
87:	ElseIf BER > (1%) and < or = (2%) then Lq = 0.17
88:	ElseIf BER > (2%) and < or = (5%) then Lq = 0.12
89:	ElseIf BER > (5%) and < or = (10%) then Lq = 0.10
90:	ElseIf BER > (10%) and < or = (15%) then Lq = 0.07
91:	ElseIf BER > (15%) and < or = (20%) then Lq = 0.04
92:	ElseIf BER > (20%) and < or = (25%) then Lq = 0.01
93:	If D < 10ms or = 10ms then d = 0.40
94:	ElseIf D > 10ms and < or = 100ms then d = 0.30
95:	ElseIf D > 100ms and < or = 500ms then d = 0.15
96:	ElseIf D > 500ms and < or = 1s then d = 0.07
97:	ElseIf D > 1s and < or = 2s then d = 0.01
98:	If B < 512Kb or = 512Kb then b = 0.05
99:	ElseIf B > 512Kb and < or = 1Mb then b = 0.05
100:	ElseIf B > 1Mb and < or = 2Mb then b = 0.04
101:	ElseIf B > 2Mb and < or = 4Mb then b = 0.03
102:	ElseIf B > 4Mb and < or = 8Mb then b = 0.02
103:	ElseIf B > 8Mb and < or = 16Mb then b = 0.01
104:	If BS < 100 or = 100 then q = 0.05
105:	ElseIf BS > 100 and < or = 500 then q = 0.04
106:	ElseIf BS > 500 and < or = 1000 then q = 0.02
107:	ElseIf BS > 1000 and < or = 5000 then q = 0.01
108:	If Cn < (1%) or = (1%) then c = 0.10
109:	ElseIf Cn > (1%) and < or = (5%) then c = 0.07
110:	ElseIf Cn > (5%) and < or = (10%) then c = 0.05
111:	ElseIf Cn > (10%) and < or =(15%) then c = 0.03
112:	ElseIf Cn > (15%) and < or = (20%) then c = 0.02
113:	ElseIf Cn > (20%) and < or =(25%) then c = 0.01
114:	SRE = ∑(b, p, d, Lq, q, c)
115:	IF SRE > THRESH_LIMIT_TYPE
116:	Then "Aggregation can be performed on both Real Time and Non-Real Time Traffic"
117:	Else "Aggregation can only be performed on Real Time Traffic"} {
118:	If PLR < (1%) or = (1%) then p = 0.40
119:	ElseIf PLR > (1%) and < or = (2%) then p = 0.35
120:	ElseIf PLR > (2%) and < or = (5%) then p = 0.22
121:	ElseIf PLR > (5%) and < or = (10%) then p = 0.14
122:	ElseIf PLR > (10%) and < or = (15%) then p = 0.06
123:	ElseIf PLR > (15%) and < or = (20%) then p = 0.04
124:	ElseIf PLR > (20%) and < or = (25%) then p = 0.01
125:	If BER < (1%) or = (1%) then Lq = 0.40
126:	ElseIf BER > (1%) and < or = (2%) then Lq = 0.35
127:	ElseIf BER > (2%) and < or = (5%) then Lq = 0.22
128:	ElseIf BER > (5%) and < or = (10%) then Lq = 0.14
129:	ElseIf BER > (10%) and < or = (15%) then Lq = 0.06
130:	ElseIf BER > (15%) and < or = (20%) then Lq = 0.04
131:	ElseIf BER > (20%) and < or = (25%) then Lq = 0.01
132:	If D < 10ms or = 10ms then d = 0.05
133:	ElseIf D > 10ms and < or = 100ms then d = 0.04
134:	ElseIf D > 100ms and < or = 500ms then d = 0.03
135:	ElseIf D > 500ms and < or = 1s then d = 0.02
136:	ElseIf D > 1s and < or = 2s then d = 0.01
137:	If B < 512Kb or = 512Kb then b = 0.05
138:	ElseIf B > 512Kb and < or = 1Mb then b = 0.05
139:	ElseIf B > 1Mb and < or = 2Mb then b = 0.04
140:	ElseIf B > 2Mb and < or = 4Mb then b = 0.03
141:	ElseIf B > 4Mb and < or = 8Mb then b = 0.02
142:	ElseIf B > 8Mb and < or = 16Mb then b = 0.01
143:	If BS < 100 or = 100 then q = 0.05
144:	ElseIf BS > 100 and < or = 500 then q = 0.04
145:	ElseIf BS > 500 and < or = 1000 then q = 0.02
146:	ElseIf BS > 1000 and < or = 5000 then q = 0.01
147:	If Cn < (1%) or = (1%) then c = 0.05
148:	ElseIf Cn > (1%) and < or = (5%) then c = 0.05
149:	ElseIf Cn > (5%) and < or = (10%) then c = 0.04
150:	ElseIf Cn > (10%) and < or =(15%) then c = 0.03
151:	ElseIf Cn > (15%) and < or =(20%) then c = 0.02
152:	ElseIf Cn > (20%) and < or =(25%) then c = 0.01
153:	SPS = ∑(b, p, d, Lq, q, c)
154:	If SPS < or = PST_1 Then
155:	"The sender can aggregate 2 numbers of packets"
156:	If SPS > PST_1 and < or = PST_2 Then
157:	"The sender can aggregate 3 numbers of packets"
158:	If SPS > PST_2 Then
159:	"The sender can aggregate 4 numbers of packets"
160:	}

The proposed algorithm has incorporated an end-to-end packet aggregation approach for voice-based packets and hop-by-hop aggregation approach for other types of data traffic. The voice packets are required to get to the destination on real-time basis and the aggregation or de-aggregation of packets takes time at each hop, when hop-by-hop is selected. Hence, the later approach is discouraged for voice-based packets. However, in case of packets bearing diverse destinations, the hop-by-hop is a crucial approach and cannot be overlooked. In case of voice packets having the same source and destination, the end-to-end approach is adopted necessarily. After making such decision, AFPA algorithm makes this distinction on packet-to-packet basis and selects a particular queue accordingly. Data packets are treated according to real time or non-real time requirement.

The maximum size of the aggregated packet cannot be more than MTU, which is determined by every network. If AADM selects to perform the aggregation, then, the level of aggregation is also adjusted according to the magnitude of PACKET_SIZE_THRESHOLD as explained earlier. The MTU for wired networks is 1500 bytes, and for IEEE 802.11 is 2300 bytes. Hence, the aggregated packet size remains below than MTU. It is therefore, in AADM the higher the PACKET_SIZE_THRESHOLD the higher would be the aggregation level and the size of aggregated packet. Besides, these thresholds are adjustable parameters, and would vary according to the specifics of the environment. Hence, it would not be wise to assume a single static threshold, suitable to any particular setting.

3.1 Selected attributes for AADM

In aggregation, there are many attributes which play vital role in defining the efficiency of a mesh network. The representative attributes are given as: congestion, delay, packet loss ratio (PLR), bit error rate (BER), buffer size, and bandwidth. In the sequel, we perform the following four aggregation decisions on the basis of aforementioned six attributes, i.e., aggregation/no aggregation; hop-by-hop/end-to-end; real time/non-real time, and packet size. For each decision we allocate weights to each of the above-mentioned attributes. Since the role of each attribute differs in each of the four decisions, their weights become changed accordingly. The list of input variables is given as: (1) bandwidth (b), (2) average delay (d), (3) packet loss ratio (Lq), (4) bit error rate (p), (5) buffer size_BS (q), (6) congestion (C). Whereas, the constants defined for the proposed algorithm are as follows: (1) THRESH_LIMIT_AGG = 40 %, (2) THRESH_LIMIT_TYPE = 50 %, (3) PST_1 = 25 %, (4) PST_2 = 50 %, (5) PST_3 = 99 %.

Table 1, further represents the notations and inputs for the AFPA algorithm. As defined that there are four decisions (D) to take, for each decision maximum weights that can be

Table 1 Notations for input parameters

Symbol	Description
SAF	Parameters sum for aggregation feasibility
SHE	Parameters sum for aggregation type selection
SRE	Parameters sum of real-time data aggregation
SPS	Parameters sum for calculation of packet size
PST	Packet size threshold

Table 2 Weights distribution schemes for attributes

Parameter	D-1	D-2	D-3	D-4
Bandwidth	0.10	0.05	0.05	0.05
Delay	0.10	0.05	0.40	0.05
PLR	0.25	0.35	0.20	0.40
BER	0.25	0.35	0.20	0.40
Buffer size	0.10	0.10	0.05	0.05

assigned to these attributes becomes different for each decision and are presented in Table 2. At first the AFPA algorithm takes the input and adapts these values according the criterion specified. These adapted or modified parameter values are added to find a sum to compare with stipulated thresholds. In this way, AFPA take useful decisions and determine the actions required to perform with packets dynamically. We come up with multiple actions serially as we traverse down by AFPA algorithm.

For decision-1, the passing criterion is 40 %, that is, if the sum of parameters is above 0.40, the aggregation would be performed; otherwise will not be considered. For decision-2, the passing criterion is 50 %, given that, if the sum of the parameters is above than 0.50, then execute end-to-end aggregation; otherwise, hop-by-hop aggregation would be performed. For decision-3, the passing criterion is again set to 50 %, in which if the sum of parameters is above that 0.50, then it will follow both the real time data and non-real time data aggregation; otherwise only the real time data traffic aggregation would be performed.

For decision-4, the MTU is specified by identifying the number of packets to be aggregated. The passing criterion tends to be progressive, as the MTU size is getting large with increasing sum. If the sum of parameters is up to 0.25, then only two packets would be aggregated; if it is above 0.25 and less than 0.50 then three packets would be aggregated, while four packets would be aggregated, if the sum is above then the 0.50 until 0.99.

3.2 Case study

A network has 2 Mb bandwidth with 10 % congestion level, 10 % packet loss ratio (PLR), 5 % bit error rate (BER), 100

ms average delay, and 500 buffer size. The prospect for performing aggregation or not in a network is as follows:

2 Mb = 0.07
10 % = 0.10
10 % = 0.10
5 % = 0.15
100 ms = 0.04
500 = 0.07

since, 0.53 (53 %), is more than 40 % so aggregation is allowed. And, the prospects for performing either hop-by-hop or end-to-end aggregation:

2 Mb = 0.07
10 % = 0.05
10 % = 0.10
5 % = 0.20
100 ms = 0.04
500 = 0.07

As 0.50 (50 %) is equal to 50 %, thus end-to-end aggregation is allowed. The prospects for performing either real time or non-real time or both:

2 Mb = 0.04
10 % = 0.05
10 % = 0.10
5 % = 0.12
100 ms = 0.30
500 = 0.04

and, 0.65 (65 %) is more than 50 %, so aggregation can be applied on real time and on non-real time data. Finally, the determined number of packets to be aggregated:

2 Mb = 0.04
10 % = 0.04
10 % = 0.14
5 % = 0.22
100 ms = 0.04
500 = 0.04

In this case here, 4 packets will be aggregated, as the calculated sum is above that 0.50 i.e., 0.52 (52 %). Nevertheless, the four hash tables illustrated in Figs. 2, 3, 4 and 5 are based on the selected six parameters. However, the mapping weight differs for each one. Our algorithm is based on the packet based aggregation, rather than frame or link based aggregation. Hence, the VoIP packet is relatively smaller than packets of other types of data. The control information in the form of header has to be sent with each packet. The small packet means more control information is being sent on the channel that leads to extra overhead. The VoIP data suffers the most
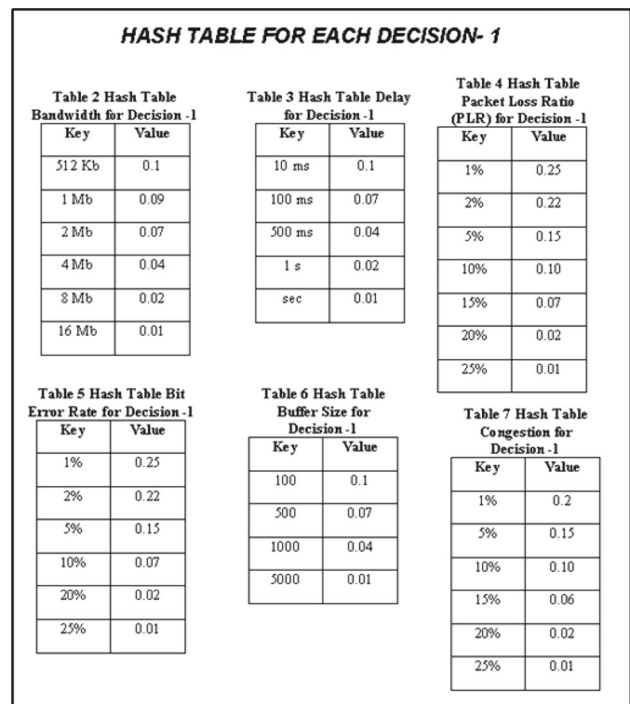

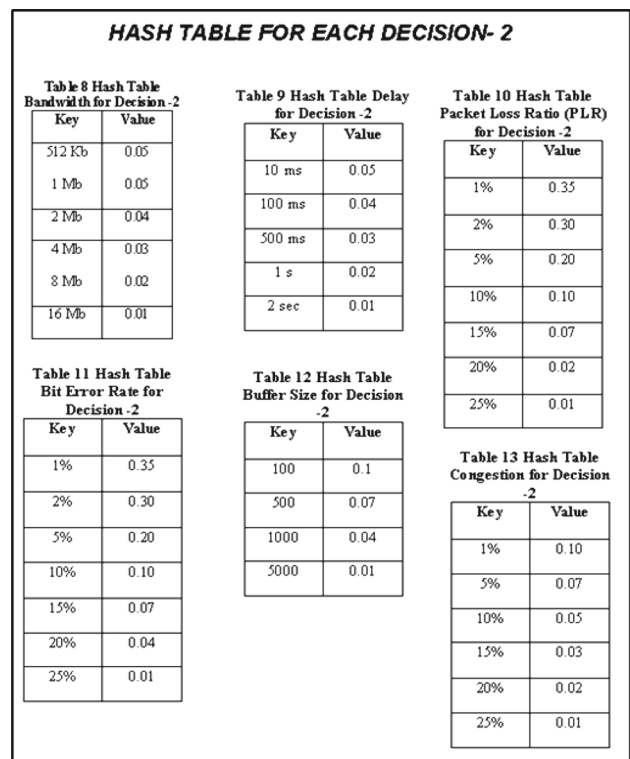**Fig. 2** Hash table of each attribute for decision- 1

Fig. 3 Hash table of each attribute for decision- 2

traditional networks. Hence the scheme is oriented towards finding the solution for VoIP streaming, which leads to the selection of End-End based approach.

HASH TABLE FOR EACH ATTRIBUTE DECISION- 3

Table 14 Hash Table Bandwidth for Decision -3

Key	Value
512 Kb	0.05
1 Mb	0.05
2 Mb	0.04
4 Mb	0.03
8 Mb	0.02
16 Mb	0.01

Table 15 Hash Table Delay for Decision -3

Key	Value
10 ms	0.40
100 ms	0.30
500 ms	0.15
1 s	0.07
2 sec	0.01

Table 16 Hash Table Packet Loss Ratio (PLR) for Decision -3

Key	Value
1%	0.20
2%	0.17
5%	0.12
10%	0.10
15%	0.07
20%	0.02
25%	0.01

Table 17 Hash Table Bit Error Rate for Decision -3

Key	Value
1%	0.20
2%	0.17
5%	0.12
10%	0.10
15%	0.07
20%	0.04
25%	0.01

Table 18 Hash Table Buffer Size for Decision -3

Key	Value
100	0.05
500	0.04
1000	0.02
5000	0.01

Table 19 Hash Table Congestion for Decision -3

Key	Value
1%	0.10
5%	0.07
10%	0.05
15%	0.03
20%	0.02
25%	0.01

Fig. 4 Hash table of each attribute for decision- 3

HASH TABLE FOR EACH ATTRIBUTE DECISION -4

Table 20 Hash Table Bandwidth for Decision -4

Key	Value
512 Kb	0.05
1 Mb	0.05
2 Mb	0.04
4 Mb	0.03
8 Mb	0.02
16 Mb	0.01

Table 21 Hash Table Delay for Decision -4

Key	Value
10 ms	0.05
100 ms	0.04
500 ms	0.03
1 s	0.02
2 sec	0.01

Table 22 Hash Table Packet Loss Ratio (PLR) for Decision -4

Key	Value
1%	0.40
2%	0.35
5%	0.22
10%	0.14
15%	0.06
20%	0.04
25%	0.01

Table 23 Hash Table Bit Error Rate for Decision -4

Key	Value
1%	0.40
2%	0.35
5%	0.22
10%	0.14
15%	0.06
20%	0.04
25%	0.01

Table 24 Hash Table Buffer Size for Decision -4

Key	Value
100	0.05
500	0.04
1000	0.02
5000	0.01

Table 25 Hash Table Congestion for Decision -4

Key	Value
1%	0.05
5%	0.05
10%	0.04
15%	0.03
20%	0.02
25%	0.01

Fig. 5 Hash table of each attribute for decision- 4

Now it is binding on the ingress node performing aggregation, to explore the ultimate and alike destinations of maximum number of packets getting through it. However, in other case the Hop-by-Hop approach would be the solution. Since single packet is not the right one candidate for aggregation i.e., if the ingress node finds only a single packet or two for some destination, it should not perform aggregation in that case and would cost the network, if it is being performed. Thus, this is one of the cases where aggregation is not going to be performed, besides the other cases we have seen in the algorithm. The algorithm takes care of different input parameters which are assigned with different suitable fuzzy logic values. Finally a computed value is obtained after summing up the fuzzy logic values and is, then compared with a threshold variable to find the feasibility of aggregation.

4 Results and analysis

This section describes simulation setup and discussion of the results observed. In the sequel, to make comparison with proposed scheme, out of the existing schemes as discussed in Sect. 2, we have selected one of the static scheme proposed by [17], due to the common application environment. We used OMNeT++ simulator to model and verify AADM. Before we discuss the simulation setup, we define throughput in our scenario first, which is the combination of three
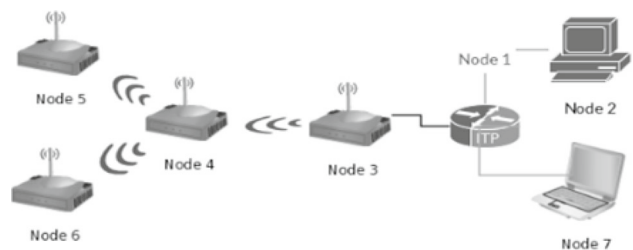

Fig. 6 Arrow topology

parameters i.e., the number of packets accurately received, total time consumed and total network load. If the throughput is increasing, then that will indicate that more packets are accurately received, less time is consumed and there is less network load.

We performed simulations of the design and development within the framework of an open scalable wireless mesh network, by using an arrow based mixed topology as shown in Fig. 6. We linked node 1 to node 6 indirectly, with the combination of wired and wireless. The node 1 is linked with nodes 2 and 3 using gigabit Ethernet (1000 Mb/s). The node 3 is connected with node 4. The former works as a gateway, while the later performs as mesh relay towards nodes 5 and 6, both inclusive.

The MTU is taken as 1500 bytes. In particular, the MAC protocol IEEE 802.11r is selected, which is designed with a special focus on VoIP. The simulation was average of ten run
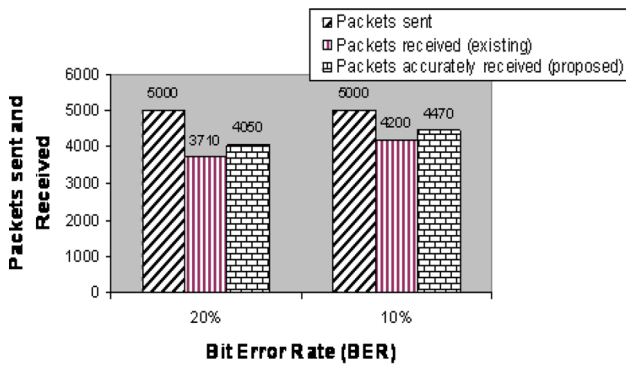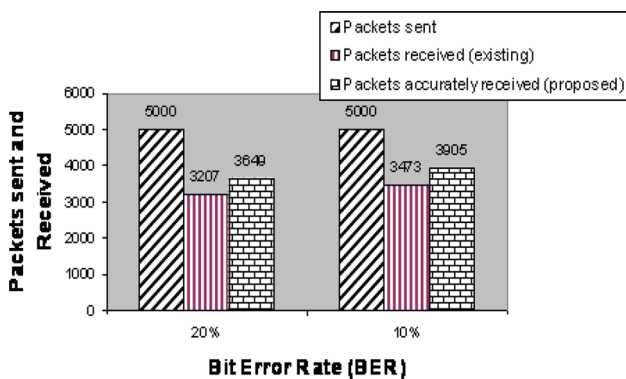
Fig. 7 Hash comparison of packets sent and received by both schemes (PLR 5 %)

Fig. 9 Throughput in existing and proposed AADM scheme (BER and PLR 10 %)
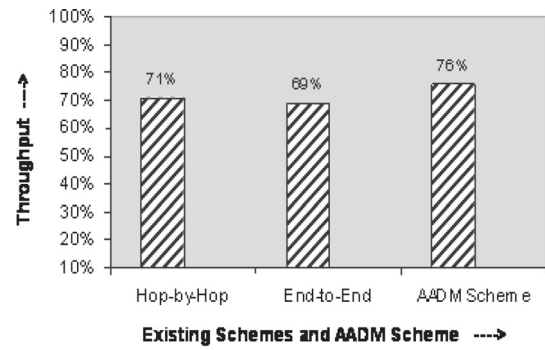
for 240 seconds each, with the variation of different traffic patterns, e.g., both constant bit rate (CBR) and adaptive bit rate (ABR). After the implementation of simulation scenario, we come up with the given results.

Figure 7, shows the comparison of packets sent and received by existing and the AADM scheme while the PLR is assumed to be 5 %. On the vertical scale the numbers of packets sent and received are shown. While on horizontal scale we used two BER levels of 20 and 10 %. As the BER gets down to 10 %, the number of packets received are increased in the same proportion for both schemes. However, the number of the packets received in the AADM scheme is higher as compared to existing scheme in both cases of the BER. Figure 8, shows the comparison of packets sent and received by existing and the AADM scheme while the PLR is set to be 15 %. On the vertical scale the numbers of packets sent and received are shown. While on the horizontal scale we have assumed two BER levels of 20 and 10 %. As the BER gets down to 10 %, number of packets received are increased in the same proportion for both schemes. How-

ever, AADM scheme outperforms in terms of the number of packets received compared to existing scheme in both cases of BER and the increment of PLR ratio.

Figure 9 shows the throughput on the vertical scale while the PLR and BER are assumed to be 10 % for both of them. On the horizontal scale the different schemes are compared. The throughput achieved by AADM scheme is proportionally higher as compared to existing hop-by-hop and end-to-end schemes. The throughput calculated for AADM amounts to 76 % in comparison with 71 and 69 % respectively for Hop-by-hop and End-to-End schemes. The higher throughput is attributed by AADM is because of its adaptive and dynamic aggregation mechanism.

Figure 10, shows throughput on the vertical scale while the PLR and BER are set to be 20 % for both. On the horizontal scale the different schemes are compared. The throughput in AADM is proportionally higher as compared to existing hop-by-hop and end-to-end schemes. Despite increasing PLR to 20 % the proposed scheme AADM has little impact on its throughput. The throughput computed for AADM amounts

Fig. 8 Comparison of packets sent and received by both schemes (PLR 15 %)
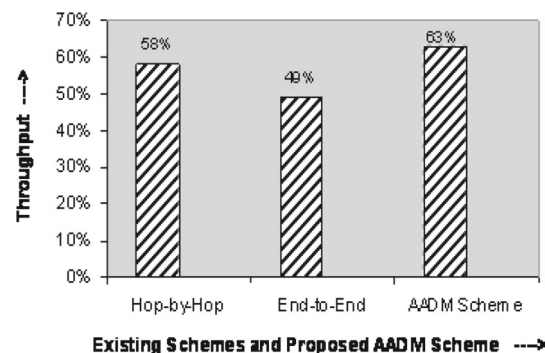
**Fig. 10** Throughput in existing and proposed AADM scheme (BER and PLR 20 %)
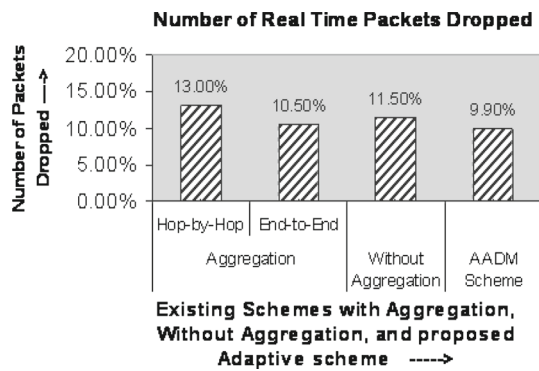
Fig. 11 Number of real time packets dropped in existing and AADM scheme

to 63 % in comparison with 49 and 53 %, respectively for the Hop-by-Hop and End-to End schemes.

Figure 11, the number of real packets being dropped is shown for AADM and existing schemes. On the horizontal scale the number of packets dropped is shown, while on vertical scale different schemes are compared for number of packets dropped. In which, the higher number of real time packets being dropped in both, the pure aggregation-based and the non-aggregation based methods as compared to the proposed AADM scheme. The number of packets being dropped in the AADM scenario amounts to 9.90 % which is the least of other ones, i.e., 13 % for Hop-by-Hop, 10.50 % for End-to-End, and 11.5 % for without aggregation.

In Fig. 12, the average delay per packet is shown for different schemes. Here we assume that BER is 5 %, PLR is 5 %, congestion is 5 % and buffer size is 10,000. On the horizontal scale, average delay has been shown in milliseconds in which, AADM outperforms in terms of average delay than that of the other schemes, which is attributed due to the less control overhead required by this scheme. Particularly under lower traffic load the AADM experiences much less delay
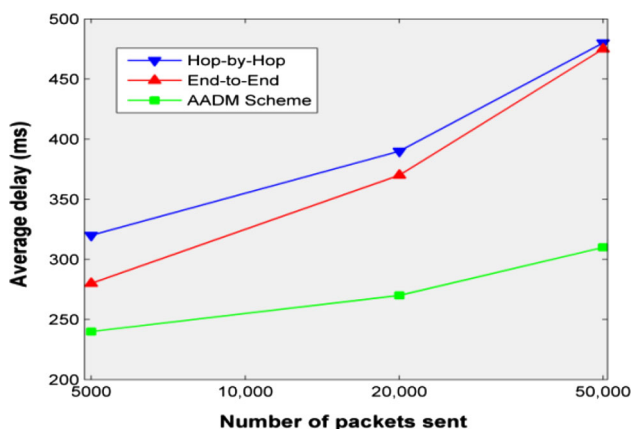
than that of the other schemes because of no aggregation, as the number of packets sent is increased; the difference increases proportionally due to the increased overhead and longer queuing delay.

5 Conclusion

In this paper, we presented a flexible and efficient aggregation framework of antifragile WMNs for application in public safety and disaster recovery. Our proposed framework not only reduces overheads associated with the large number of small packets but also conserves useful resources such as bandwidth and power thereby, in part, creates ample space for its application in a PSDR setup. We focused on core parameters such as buffer size, link quality, bandwidth, delay, congestion and packet loss, to process and identify the likelihood of aggregation. We developed a novel algorithm that makes four crucial decisions with respect to aggregation and its associated options. It directs routing algorithm what to do in a particular scenario, as a different strategy is required to cope with a different situation.

We have adopted a packet level aggregation to minimize the negative effect of scarcely available resources indicated by the input parameters to the algorithm. AADM introduces scalability, better performance, power consumption optimization, network efficiency and QoS. It reduces the overall number of packets in the mesh, and thus lessens multi-hop contention and packet loss due to collisions. Our research has been supported by OMNeT++ based simulation results, which shows that the AADM model optimizes the current scenario by making useful decisions and exploits the available resources in an optimum manner. A possible future direction is to concentrate another potential benefit of AADM in terms of energy consumption, and more performance measures can be taken into account that have not been addressed by this study.

Fig. 12 Average delay per packet (BER 5 %, PLR 5 %, congestion 5 %, buffer size 10000)

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