A Cross-Layer On-Demand Routing Protocol for Delay-Sensitive Applications

Lamia Romdhani and Christian Bonnet
Institut Eurécom - {romdhani, bonnet}@eurecom.fr

***Abstract*— Nowadays, the cross-layer design approach is an important concept in mobile ad-hoc networks which is adopted to solve several open issues. It aims to overcome MANET performance problems by allowing protocols belonging to different layers to cooperate and share network status informations while still maintaining separated layers. Indeed, the mechanisms on how to access the radio channel are extremely important in order to guarantee QoS and improve application performance.**

In this work, we present a Cross Layer routing protocol called CLAE which is based on the cooperation between the On-Demand AODV routing protocol and the new IEEE 802.11e MAC protocol (EDCA). This proposal aims to find the best path according to application requirements in terms of delay, bandwidth, route stability, etc. Without loss of generality, this paper focuses only on determining the path with the lowest delay. Each node periodically estimates the average transmission delay for each class of service defined by 802.11e. This information is injected into routing requests and replies crossing each node. The sender is then able to select the best path which fits its delay requirement. Furthermore, in order to overcome transient network characteristics due to new communications set up and mobility, we develop a new buffer management scheme for the audio class of service that aims to discriminate audio packets according to their tolerated end-to-end transfer delay and their current experienced delay. The simulation results demonstrate that our proposal improves the performance of delay-sensitive applications while maintaining a good packet delivery ratio of other traffics.

Keywords: mobile ad hoc networks, cross-layer design, routing protocol, quality of service, delay-sensitive applications.

I. Introduction

A Mobile Ad-hoc Network is a set of wireless mobile nodes dynamically forming a temporary network. The goal of this architecture is to provide communication facilities between end-users without any centralized infrastructure. In such a network, each mobile node operates not only as a host, but also as a router.

Quality of Service (QoS) support is critical to wireless home networking, video on demand, audio on demand and real-time voice IP applications. Time-bounded services such as audio and video conference typically require some specified bandwidth, delay and jitter guarantee, but can tolerate some losses. One of the main problems of wireless links is that all the nodes compete for the resources and channel without taking into account knowledge about neighbor communications. There is not any consideration to guarantee packet delay and jitter to flows supporting time-bounded multimedia services.

Enhanced Distributed Channel Access (EDCA) is a contention-based Hybrid Coordination Function (HCF) channel access specified in IEEE 802.11e [4], [6]. The goal of this scheme is to enhance the Distributed Coordination Function (DCF) access mechanism of IEEE 802.11 [5] and to provide a distributed access approach that can support service differentiation. The proposed scheme provides capability for up to four types of traffic classes. It assigns a short CW (Contention Window) to classes that should have higher priority in order to ensure that in most cases, higher-priority classes will be able to transmit before the lower-priority ones. Indeed, the CW_{min} parameter can be set differently for different priority classes, yielding higher priority classes with smaller CW_{min}. For further differentiation, in 802.11e different IFS (Inter Frame Space) can be used according to traffic classes. Each Traffic Category (TC) within the station behaves like a virtual station: it contends for access to the medium and independently starts its backoff timer according the basic scheme algorithm.

Despite many enhancement mechanisms that have been introduced based EDCA, to achieve QoS support, performance evaluation results in multi-hop networks show that EDCA still suffers from significant throughput degradation and high delay at high load conditions. They are caused by the increasing time used for channel access negotiation and transit network characteristics. In this context, the route quality plays a very important role in the success of application delivery and QoS support. To this end, we address interactions between the EDCA MAC scheme and AODV routing protocol. The basics of the proposed scheme is to estimate the transmission time at each node and for each class of service. In this paper, when trying to find a route from a source to destination, the destination has to send a Route REPly (RREP) message for each received Route REQuest (RREQ) message. A delay field is added to the RREP message, which is updated each time this message goes through an intermediate node. Moreover, in order to overcome transient network characteristics due to new communications set up and mobility, we develop a new buffer management scheme for the audio class of service that aims to discriminate audio packets according to their tolerated end-to-end transfer delay and their current experienced delay.

The remainder of this paper is organized as follows. In Section II, we review the most important works that have been done so far to enhance MANET performance through the use of cross layer architecture. In Section III, our research work's motivations are given. The description of the proposed CLAE cross-layer model is given in Section IV. Simulation methodology and performance evaluation of our proposal are detailed in Section V. Section VI concludes the paper by summarizing results and outlining future works.

II. Related works

In this section we review the most related works close to our proposal and that address cooperation between MAC and routing protocols.

In [3], the authors proposed a mechanism for detecting network congestion, in order to improve the performance of all types of traffic. Indeed, there are two metrics which are used to measure the congestion level. The first one, is the average MAC layer utilization around each node. Instantaneously, this metric can be equal to 1 or 0. It is equal to 1 if the MAC layer is utilized. The second metric, is the instantaneous interface queue length. The routing protocol looks to establish routes over no congested nodes. However, if we avoid busy nodes in route establishment, there are some routes that cannot be established even if they exist. At higher layer, these metrics can be used to decide whether or not data compression. Indeed, when the medium is busy the sender can decide to compress the data. However, the compression should represent a trade-off between bandwidth consumption and the CPU time used for compression and decompression. [2] investigated a local interaction among protocols in a MANET node. For example, the MAC layer exploits the topology information collected by the network layer to achieve fair channel scheduling and fix the problem related to hidden and exposed terminals. An enhanced backoff scheme is introduced. The authors suppose that a node has a knowledge of the whole network topology and so a proactive routing protocol should be used. Hence, it seems that for some scenarios, it is very hard, costly, and not efficient to address this cross layer architecture regarding the dynamic traffic nature and the high mobile node speed. Moreover, no information has been provided on how to compute the path per-formability index or other cross layer parameters.

In [10], a study of cross-layer design based on power conservation and congestion informations in ad-hoc networks has been presented. The authors described a power control based cross layer architecture. Indeed, they detail the significant impact of power control on all layers above the physical layer. Furthermore, they summarize several works that address power saving in the protocol stack and show how the power information could be considered at each layer. Moreover, the work claims that exchanging topology information between different layers, through their interfaces, is very important to support QoS. The exchanged parameters could include geometric location, channel and link conditions. A proposed mechanism, that uses the number of neighbors around the node to adjust transmission power, has been presented.

III. Motivations

In the AODV (Ad hoc On-Demand Distance Vector) routing protocol, a minimum hops algorithm is applied to establish routes between sources and destinations [11]. Our work considers another metric in the route establishment process. This metric, which is included in route cost computation takes into account the MAC layer average delay transmission of all nodes participating in the route. We believe that by doing this, we are able to provide some how a distributed load balancing among nodes by choosing nodes that are less congested and hence have lower delay. Indeed, the first received RREP in the basic AODV does not correspond usually to the one experiencing the lowest delay, as routing control packets have always the highest priority and are enqueued at the head of the queue of the MAC layer. However, mobile ad hoc networks have dynamic quality characteristics. So, the cost of the established route could change fast which is not useful for delay-sensitive applications as we can provide a bounded end-to-end delay. To this end, our proposal aims to select the best route according to the estimated end-to-end delay and not to the number of hops. Moreover, it suggests that the MAC layer of each intermediate node observes the experiencing delay of each ongoing packet and gets from the routing layer the required remaining time after which this packet has to reach the destination. Considering these parameters, we address a buffer management scheme. The aim of this scheduling process is to allow packets to be delivered within maximum end-to-end delay.

This is what will be described in the following section. Our mechanism is based on cross-layer interaction between MAC and routing protocols. Indeed, even EDCA is introduced to support link layer service differentiation, we show that it is not enough to provide a QoS guarantees for real-time applications specially in dynamic networks. Hence, sharing informations between MAC and routing protocols, is very important to achieve a good performance and overcome a low packet delivery ratio in a network with frequently changing characteristics.

IV. Proposal description

As we mentioned above, our cross-layer proposal is based on the interaction between MAC and routing layers. Hereafter, we detail the proposed mechanisms that have to be considered in this new architecture.

A. Estimating node's transmission delay

Several research works [9] have showed the importance of considering the impact of MAC-layer on the application performance. In our scheme, when the MAC layer receives unicast packet from the routing layer to be sent to the next-hop, it saves the current time. When it receives later the ACK for this packet, it computes the transmission delay which is simply the ACK's reception time by the MAC layer minus the packet's reception time from the routing layer. This delay takes into account queuing, transmission, and propagation delays, and is computed for all transmitted packets during a configurable *period* called T. If there is no traffic, only propagation delay is considered. At the end of each period T, the node updates the average transmission delay for the corresponding priority. The obtained time also takes into account the eventual retransmission retries at the MAC layer. Furthermore, only successfully sent packets are considered for estimating the average transmission time in intermediate nodes. We note by D^j_{curr} the current computed average packet delay at step j.

To minimize the bias against transient delay, we use an estimator of Exponentially Weighted Moving Average (EWMA) to smooth the estimated values. Let D^j_{avg} be the average delay at step j (for each update period T) computed according to the following iterative relationship:

$$D^j_{avg} = (1-\alpha) * D^j_{curr} + \alpha * D^{j-1}_{avg} \quad (1)$$

where j refers to the j^{th} update period T and D^j_{curr} is the instantaneous delay, α is a smoothing factor and it effectively determines the memory size used in the averaging process. In order to take into account the network dynamics, we choose α to be the current Medium Utilization (MU) measured around the node during the previous period. The MU metric is computed as follows:

$$MU = \frac{T-IdleTime}{T}$$

Where IdleTime is the portion of time when the medium is idle. This leads to α in the interval $[0, 1]$. We choose the using of the MU metric, since it quantifies the up-to-date state of the wireless medium and takes into account neighbor's transmissions and deferring backoff times.

B. Including delay information in routing control packets

The routes are established based on the end-to-end estimated delay transmission cost as computed above. Each RREQ packet includes the traversed path cost. Then each node maintains for each reverse route this cost in the routing table so that routes are built based on the route's cost, defined as follows:

$$Cost_{(s,d)} = \sum_{c \in path(s,d)} D_{avg}$$

Before sending a route reply, the destination has to include the computed route's cost which is mentioned in the routing request that it received.

Note that in the basic AODV, intermediate nodes can send back to the sender routing reply message when they have already stored information about routes to each the destination. In our proposal, we disable this feature as we want the sender to receive an up-to-date information about the estimated end-to-end delay. Hence, all route replies are sent by the destination.

C. Selecting the best path

As described above, the basic AODV routing protocol uses the minimum hop count critirea to establish routes between sources and destinations. However, considering the new cross-layer model, we follow a new route discovery scheme. Indeed, the source uses the first path retrieved while still accepting other routing replies during the user session lifetime. When a new routing reply arrives, the source observes the estimated delay included in this reply. If it is less than the stored delay of the current available route, it updates the metric of that route entry and uses it for the next packets. This policy allows to reduce the time that the source waits to send the data packets as it cannot know in advance how many route replies it will receive.

D. MAC layer buffer management interaction with routing layer

The dynamic characteristics of mobile ad-hoc networks decrease the efficiency of the chosen route. In order to overcome transit network characteristics, using these metrics in a cross-layer model might be inefficient because sometimes they are based on inaccurate values which do not reflect the real situation around a given node. Moreover, since a node moves with an arbitrary speed and toward an arbitrary destination, the computed metrics (according to the average MAC layer delay, or to the participation of the node in communication and the traffic load level around it) could change over time. Therefore, other nodes that consider the metrics of that node to build routes, could have an inaccurate information since they change according to mobility, traffic, and capacity. To overcome this problem, we develop a new buffer management scheme for the audio class of service that aims to discriminate audio data packets regarding their current experienced delay and their tolerated end-to-end transfer delay (*400ms in this context* [14]). The mechanism consists of reordering audio packet transmissions in the queue according to two parameters. The first, is the total delay that packet experienced during the route until arriving at the current node. To avoid synchronization problems, we incorporate in the packet header the sum of times that the packet experienced in the previous intermediate nodes. The second parameter is the estimated delay that the packet has to experience before it reaches the destination. This parameter is available in the routing table and it is shared with the MAC layer. If the sum of the two delays close to the maximum tolerated end-to-end delay, the packet has to be enqueued at the head of the audio class'queue. Audio packets are then inversly ordered according to their remaining lifetime. Moreover, packets that exceed their maximum tolerate end-to-end delay are dropped. By this way we alleviate the network congestion since these packets will not be considered at the destinations.

V. Performance evaluation

We implemented our proposal in the ns-2 network simulator [7]. We have extended the AODV protocol and EDCA scheme to support our cross-layer algorithm. We report in this section the results of simulations we have done for various network scenarios. We also provide an analysis of the obtained performance results.

A. Scenario description

The simulated scenarios consist of 50 nodes located in a uniform distribution within an area of 1500x300m forming a multi-hop network. These scenarios are generated by the enhanced random way-point mobility model [13]. Our simulation uses different types of traffic to evaluate service differentiation. Three queues are used in each active station. The highest priority queue in each station generates packets with packet size equal to 160 bytes and inter-packet interval of 20 ms, which corresponds to 64 Kbit/s PCM audio flow (**high sensitive-delay applications**). The medium traffic queue generates packets of size equal to 1280 bytes each 10 ms which corresponds to an overall sending rate of 1024 Kbit/s. The low priority queue in each station generates packets with sending rate equal to 260 Kbit/s, using a 200 bytes packet size. To increase the load of the system, we gradually increase the number of flows until 84 flows: 30 audio flows, 25 video flows, and 29 best effort flows. Note that the number of source nodes is 30 sources. Moreover, we consider an arbitrary starting and end time of communications to show how the proposed model could be adapted to the dynamic network load. The radio model is very similar to the first generation WaveLAN radios with nominal radio range of 250m. The nominal bit rate is 2 Mbps. In our simulation the nodes move at an average speed of $20m/s$.

TABLE I

IEEE 802.11A PHY/MAC PARAMETERS USED IN SIMULATION

SIFS	$16\mu s$
DIFS	$34\mu s$
ACK size	14 bytes
Data rate	36 Mbits/s
Slot_time	9 μs
CCA Time	$3\mu s$
MAC Header	28 bytes
Modulation	16-QAM
Preamble Length	$20\mu s$
RxTxTurnaround Time	$1\mu s$
PLCP header Length	$4\mu s$

In the following simulations, we assume that each wireless station operates at IEEE 802.11a PHY mode-6, see network parameters shown in Table I.

Table II shows the network parameters selected for the three TCs.

We have done an extensive set of simulations to observe the effect of the update period T on the delay performance. In the following simulations we have chosen T equal to 2 seconds, which provides us a good latency. Determining dynamically the optimal value of T is out of the scope of this paper.

B. Performance metrics

To evaluate the performance of the different schemes, the following metrics are used:

- **Latency distribution:** latency distribution allows to trace the percentage of packets that have latency less than the maximum delay required by the applications. Real-time flows require both low average delay and bounded delay jitter. So we will also use the two metrics of latency distribution and delay variation.
- **Throughput**: this metric shows the total number of bytes that have been successfully delivered to the destination nodes.

TABLE II

MAC PARAMETERS FOR THE THREE TCS.

Parameters	High	Medium	Low
CW_{min}	7	31	31
CW_{max}	15	31	1023
AIFS(μs)	34	43	52
Packet Size$(bytes)$	160	1280	200
Packet Interval(ms)	20	10	12.5
Sending rate$(Kbit/s)$	64	1024	128
slot-time(ms)	16us	16us	16us

C. Performance results and analysis

We present in this subsection the performance of the basic **AE (AODV-EDCA)** and **CLAE** for the various metrics presented above. We simulate 10 random scenarios and then take the average of the performance values.

Hereafter, we present results of different network mobility scenarios. In Figure 1 and Figure 2 we show the delay distribution of audio traffic and the variation of throughput over the simulation time respectively. The simulated scenario corresponds to a pause time of node equal to zero (in the random waypoint model), that leads to a highly mobile network. However, Figure 3 and Figure 4 are obtained in case of a static ad hoc network (no mobility - infinite pause time).

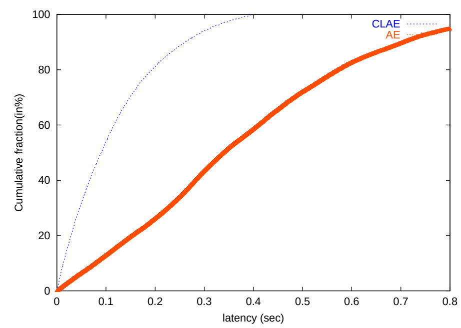

Fig. 1. Delay distribution of CLAE and AE for a highly mobile ad hoc network (pause time = 0)

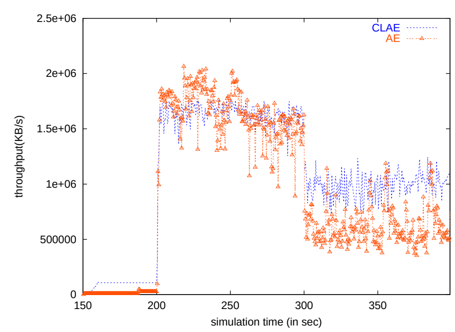

Fig. 2. Throughput of CLAE and AE for a highly mobile network (pause time =0)

We show the latency distribution for audio traffic in Figure 1 and Figure 3 in which a number of 30 voice flows is used to show the delay performance. On a cumulative distribution plot, an ideal result would coincide with the y-axis, representing 100% of results with zero latency. Although we cannot reasonably expect zero latency, we would like to obtain a consistent performance, corresponding to a vertical line.

Observing Figure 1 and Figure 3, we can conclude that mobility affects the delay performance of both CLAE and AE due to the often route failures. However, CLAE always outperforms the basic AE. There are considerable differences between them, i.e. more than 98% of audio packets for our

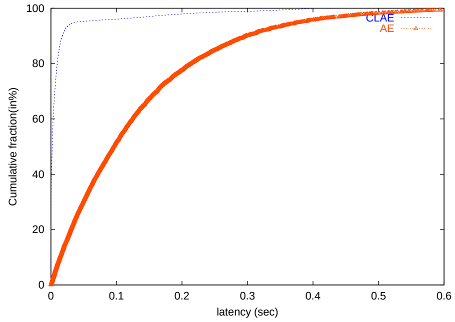

Fig. 3. Delay distribution of CLAE and AE for a static ad hoc network (no mobility)

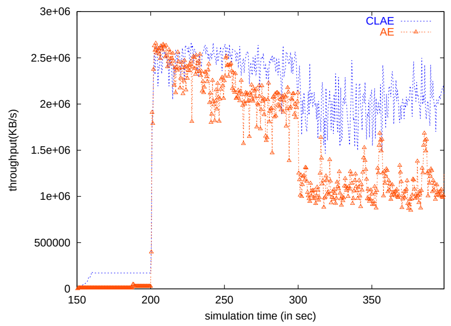

Fig. 4. Throughput of CLAE and AE for a static ad hoc network (no mobility)

model have delay less than $400ms$, whereas only 60% of audio packets for the layered protocols have delay less than $400ms$.

These results are due to the good effect of including MAC layer delay in routing process and the importance of introducing the buffer management mechanism to satisfy application requirements. Note that the maximum delay of audio packets of CLAE is less than $400ms$, whereas for the basic mechanism, the maximum value is much more than $800ms$. Indeed, getting a delay more than $400ms$ degrade the quality of audio reception. This is the effect of our proposal that drops packets which have experienced more than the maximum tolerated delay before reaching destination.

Choosing routes with minimum delay will decrease the number of congested nodes that build a QoS route. This allows load balancing in the network. Furthermore, the dropped packets in the intermediate nodes decrease the congestion of the network. Indeed, there are less competing packets to gain the medium in the next hops, that these packets expect to traverse, and so more successful transmissions. Figures 2 and 4 demonstrate the good effects of the considered schemes. Indeed, the total delivered throughput obtained by CLAE is higher than AE. This ensures a good delivery packets for video and best effort traffics.

Obviously, when nodes are static, the improvement in term of throughput (Figure 4)is much better than that obtained in mobile network (Figure 2). The results are proved by the fact that routes are more often re-established due the the high mobility. So, less data packets are transmitted.

From the simulations, we can conclude that CLAE can provide a good enhancement in term of delay for delay-sensitive application while achieving a good delivery packets for other traffic categories. Due to space limitations, we cannot include more simulations that we have conducted but we believe that performance results presented here are quite representative of what we have obtained.

VI. Conclusion and Future Works

This paper presents a new MAC and routing cross-layer approach called CLAE. The cooperation goal aims to provide a good delay and delay jitter for delay-sensitive applications but it could also be applied for other QoS parameters such as bandwidth, loss rate, and a route stability. The approach is an adaptive service differentiation based on buffer management and route establishment strategy. Performance evaluation using ns-2 simulator shows the importance of considering the MAC-layer delays in route selection process. Overall, we conclude that our mechanism demonstrates significant benefits at high and unstable traffic scenarios. Even though we implemented the model in AODV, the technique used is very generic and can be used with any on-demand protocol such as DSR. Furthermore, this proposal can be applied to single channel and multi-channel based medium access protocols, and there is no need for synchronization. Additionally, the scheme could also be applied for the basic 802.11 that does support service differentiation at the link layer.

Important future work is to evaluate the performance of this proposal for other parameters such as bandwidth and route stability. We also plan to study its performance when using another on-demand routing protocol such as DSR or even an extended version of reactive routing protocol such as OLSR by including of the required information in the TC (Traffic Control) control messages. Doing real experimentations, at least for static ad hoc networks, is also to be considered.

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