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Original Article

Optimizing Real-Time Voice Quality in Ad Hoc Wireless Networks: A Knapsack-Enhanced Dynamic MAC Protocol

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Abstract *- This work aims to address the challenges of providing Quality of Service in real-time data transmission over mobile ad hoc networks. The main focus is to address the MANETs constraints and complexities that arise due to its characteristics regarding quality-of-service provision. The work proposes an optimized dynamic audio session mapping MAC mechanism to improve the voice quality in multi-hop MANETs. The proposed mechanism environment to overcome the limitations of the existing Dynamic Session Mapping scheme regarding voice quality provision. The proposed algorithm dynamically maps the voice session based on the quality of the session, computed using packet loss and end-to-end delay. The approach improves the existing MAC protocol session mapping by employing an optimized intelligent session mapping with the assistance of the knapsack algorithm. It dynamically allocates voice sessions to maximize the cumulative rating factor within network constraints. The effectiveness of the proposed methodology is assessed using NS2 simulations, revealing notable improvements such as reduced packet loss, enhanced packet delivery, and minimized delays. Comparative analyses against the Dynamic Session Mapping scheme under various traffic conditions highlight the efficacy of the proposed work in improving real-time voice session throughput. This work contributes to advancing the capabilities of MANETs, particularly in supporting high-quality real-time communication applications such as Voice over Internet Protocol.*

Keywords *- MANETs, QoS, MAC, VoIP, Knapsack algorithm.*

1. Introduction

Wireless communication has undergone a significant transformation with the introduction of MANETs. This network plays an important role when setting up a traditional network is impractical, such as during disaster recovery, military operations, and law enforcement. However, achieving good quality in real-time applications is challenging in MANETs due to its characteristics, such as adaptability, mobility, and multi-hop communication, particularly achieving good quality voice communication like Voice over Internet Protocol (VoIP).

The existing 802.11e Medium Access Control (MAC) protocol is less effective in MANETs due to its infrastructure-less nature and resource constraints. Further, a significant issue is the delay in voice communication, i.e., the delay from the speaker's mouth to the listener's ear, made even more complicated by the limited resources of MANETs.

MANETs play a crucial role in scenarios where traditional infrastructure is not feasible due to the absence of a predefined network infrastructure [1]. Therefore, there is a need for innovative solutions and the design of protocols to address the challenges posed by the distinctive characteristics of MANETs and to achieve quality in voice communication. Ongoing efforts focus on creating adaptive Quality of Service (QoS) mechanisms that can effectively operate in the MANET environment, particularly in infrastructure-less and dynamic, resource-constrained MANET settings. The primary goal of these developments is to minimize mouth-to-ear delay to achieve high-quality voice communication.

The work aims to improve the quality of voice communication in MANET. It suggests a novel way of managing voice sessions between communicating entities in a MANETs environment. The decision on voice session mapping is scheduled based on important factors such as packet loss and mouth-to-ear delay of the voice session. The work is particularly designed to enhance the quality of voice in a multi-hop environment, and it also highlights issues with existing methods of ensuring good Quality of Service (QoS). Furthermore, we use an optimization method to improve the quality of voice sessions significantly, benefiting applications like Voice over Internet Protocol (VoIP) that require good quality and real-time communication [2].

2. Background

As per the 802.11 standard, wireless communication networks have two main ways to control how users access the communication medium: the contention-based Distributed Coordination Function (DCF) and the centralized Point Coordination Function (PCF). DCF operates on the concept of Carrier Sense Multiple Access with Collision Avoidance.

This means a communication entity checks if the channel is free before sending information through the medium. If it is, the source can transmit. This method works well in wireless infrastructure environments, where all communicating entities have equal priority to access the medium. PCF, on the other hand, manages the medium through a coordinator. It's a centralized approach to handling the communication medium [3].

The existing standards are enhanced by introducing 802.11e, extending the Distributed Coordination Function (DCF) with a novel standard called Enhanced Distributed Channel Access (EDCA). EDCA is developed to provide varying levels of service for different traffic types, categorizing them into four categories: 1) AC_BE for best-effort, 2) AC_VI for voice, 3) AC_VO for video, and 4) AC_BK for background traffic.

All categories have their own set of parameters, including the Contention Window (CW_{min} and CW_{max}), Arbitration Inter-Frame Space (AIFS), and retry limits [4]. These parameters are customized for specific types of traffic, allowing for precise and efficient control of the communication medium. EDCA ensures that different types of traffic receive the required attention and priority. This, in turn, enhances the overall Quality of Service (QoS) in wireless networks.

Careful consideration is needed to achieve good throughput and QoS for real-time voice communication over the internet, like VoIP. Voice communication involves several steps:

1. Turning analog signals into digital data.
2. Compression of the digital data.
3. Encoding it using voice codecs with specific data rates.
4. Dividing the resulting data into packets with constant bit rates.
5. Sending these packets at regular intervals.

These packets are framed and transmitted at the Medium Access Control (MAC) layer. During communication, various challenges arise, particularly variations in the time it takes for the packets to arrive, which can be obstacles to achieving quality voice communication.

Ensuring voice communication quality in MANETs is more complex since nodes are connected without a fixed infrastructure. Moreover, multi-hop communication introduces dynamic delays, significantly impacting the mouth-to-ear delay in voice communication. MANET adds an extra layer of complexity with its peer-to-peer multi-hop connections and adaptable, self-forming capabilities, making it challenging to ensure top-notch voice communication quality.

The primary concern for achieving quality in voice communication in MANETs is intricately tied to the concept of mouth-to-ear delay, representing the time it requires for voice information to travel between communication entities. MANETs experience dynamic delays as communication happens in a multi-hop manner, where entities rely on intermediate nodes [5].

This work addresses the challenges faced in achieving quality voice communication in MANETs that use the 802.11e standard with multi-hop communication. The 802.11e standard provides EDCA and offers a framework with a dedicated access category specifically designed for voice communication, known as AC_VO.

The primary aim is to enhance voice communication quality by developing an optimized dynamic Medium Access Control (MAC) protocol and tackling challenges arising from resource constraints and dynamic conditions in MANETs. Moreover, it must address issues such as packet loss, delays in packet delivery, and other inefficiencies that may be present when compared to existing dynamic voice session mapping algorithms [6].

The proposed approach is specifically designed for 802.11e-based MANETs, and it is designed with the help of a sophisticated optimization algorithm, i.e., knapsack algorithm, to strategically allocate voice sessions to specific medium access categories (AC_VO_1, AC_VO_2, and AC_VO_3). It computes the rating factor for each voice session at regular intervals. This rating factor is determined based on dynamic characteristics such as mouth-to-ear delay and packet loss of voice sessions.

The algorithm optimizes and prioritizes voice sessions according to the computed dynamic factors. Following the rating computation, the knapsack algorithm intelligently selects voice sessions to maximize the cumulative rating factor while adhering to network constraints. In other words, it chooses the combination of voice sessions that, when placed in the available medium access categories, collectively yields the highest overall rating, all within the limitations set by the network [7]. This paper addresses the challenges faced in achieving voice communication quality in MANETs that use the 802.11e standard for multi-hop communication. The primary focus is enhancing voice communication quality by designing an optimized dynamic Medium Access Control (MAC) protocol.

The subsequent sections of the research delve into the proposed optimized dynamic session mapping MAC protocol, which introduces a knapsack-like approach for dynamic session mapping. To assess the effectiveness of the proposed method, we carried out extensive simulations utilizing the Network Simulator 2 (NS-2). This simulator allows users to mimic and analyze how the proposed protocol would work in various scenarios. It provides valuable insights into its effectiveness in improving voice communication quality in multi-hop MANETs.

The simulated network topology incorporated a MANET infrastructure-less mode environment, capable of handling diverse traffic types, including background, best effort, and Voice over Internet Protocol (VoIP) calls using the G.711 codec. Throughput computations, focusing on voice traffic, were meticulously conducted. The impact of the proposed work on voice session throughput is visually represented through detailed graphs.

Comparative analyses were performed, contrasting the existing dynamic voice session mapping scheme against the proposed method under various traffic conditions, namely background and best effort. Finally, this work helps improve MANETs, particularly for voice communication, i.e., talking on the phone over the internet (VoIP). It designs an effective solution and special way of deciding how nodes in MANET should talk to each other. The aim is to make sure the quality of phone calls is better in these dynamic networks where resources are constrained [8].

3. Optimized Dynamic Session Mapping Mechanism

To overcome challenges in ensuring smooth voice communication quality in multi-hop MANETs, this section suggests a refined dynamic Medium Access Control (MAC) protocol designed specifically for effective network adaptation. In response to the challenges of achieving QoS for voice communication in multi-hop MANETs, this section proposes an optimized dynamic Medium Access Control (MAC) protocol particularly designed for network adaptation.

The objective is to enhance voice communication quality in MANETs by addressing resource constraints, network characteristics, and dynamic conditions. It optimizes voice communication quality by reducing packet loss, improving packet delivery, and minimizing delays. The aim is to enable efficient and reliable communication, especially for real-time applications like VOIP.

The major innovation in the proposed ‘Optimized Dynamic Session Mapping’ MAC protocol is designed to prioritize voice sessions based on specific quality metrics of voice sessions. This quality metric is computed by a rating factor, considering both mouth-to-ear delay I_{delay} and packet loss I_{loss}, and it is shown in Equation 1 [2, 3].

Further, the Knapsack algorithm is used for intelligent session mapping and dynamically allocates voice sessions to suitable medium Access Categories (ACs) within the 802.11e-based MANET environment. The concept of computing the R factor is inspired by the standard put forth by International Telecommunication Union (ITU). This standard provides guidelines for objectively evaluating the quality of coded speech signals transmitted through the telephone network.

$$R_{factor} = R_{max} - I_{delay} - I_{loss} \tag{1}$$

The R factor provides a rough estimate of voice quality, as per ITU-T recommendation, and serves as a measure to determine the priority of packets. In this context, the R factor spans a scale of 0 to 100, where a higher score signifies superior quality (100 representing the highest and 0 the lowest).

Typically, values below 50 are regarded as unacceptable, those falling between 90-100 are deemed highly satisfying, and the range of 80-90 is considered satisfactory. Users may find values between 70 and 80 acceptable. One can prioritize voice session packets based on their R factor score; those with an R factor above 90 get lower priority, those between 80 and 90 receive medium priority, and those between 50 and 80 are assigned higher priority [10].

In our work, the 802.11e AC architecture, the AC-VO (Access Category for Voice), is divided into three sub-access categories, each designed for specific types of voice communication. These sub-access categories are named as follows:

AC_VO_1 (High Priority Voice): This sub-access category has the highest priority among voice spells. When AC_VO gains control of the channel, AC_VO_1 gets the first preference.

AC_VO_2 (Medium Priority Voice): This sub-access category is designed for voice spells with medium priority. If the channel is available after AC_VO, AC_VO_2 gets access next in line.

AC_VO_3 (Low Priority Voice): AC_VO_3 is the sub-access category with the lowest priority among voice spells. It gets access to the channel after AC_VO and AC_VO_2.

The priority order for accessing the medium is AC_VO_1, taking precedence, followed by AC_VO_2, and subsequently AC_VO_3.

3.1. Knapsack Algorithm

The algorithm efficiently determines the best combination of voice sessions to be included in the knapsack to maximize the cumulative rating factor while considering the capacity constraint. It employs dynamic programming to iteratively compute and update values in the table, leading to an optimized solution.

Thus, the knapsack algorithm intelligently allocates voice sessions to the appropriate medium access subcategories AC_VO_1, AC_VO_2, and AC_VO_3 while maximizing the cumulative rating factor. The algorithm selects a subset of voice sessions based on specific weights, i.e., rating factors and values, i.e., priority levels, while staying within the capacity constraint, i.e., medium access capability [11].

1. The algorithm initializes a table for dynamic programming to optimize voice session selection based on rating factors, priority levels, and medium access capability.
2. It iteratively fills the table by considering each voice session and its impact on maximizing the cumulative rating factor within the given medium access capability.
3. The result is a table representing the optimal selection of voice sessions for intelligent mapping to medium access categories in a network, extending overall efficiency.

```
Algorithm-1
1. Initialization:
        def knapsack_algorithm(sessions, capacities):
        n = len(sessions)
        dp = [[0] * (capacities + 1) for _ in range(n + 1)]
2. Dynamic Programming Loop:
        for i in range(1, n + 1):
        for w in range(capacities + 1):
3. Check Weight Condition:
        if sessions[i - 1].weight <= w:
4. Update Dynamic Programming Table:
        dp[i][w] = max(
        sessions[i - 1].value + dp[i - 1][w - sessions[i -
        1].weight],
        dp[i - 1][w],
        )
5. Handle Non-Matching Weight:
        else:  dp[i][w] = dp[i - 1][w]
6. Return Dynamic Programming Table:
        else:  dp[i][w] = dp[i - 1][w]
```

Algorithm 1 systematically calculates the optimized subset of voice sessions to maximize the cumulative rating factor by staying within the capacity medium access capability. It enables a dynamic programming method and iteratively updates a table (dp) to store maximum values. This output plays a major role in intelligently allocating the voice sessions to medium access subcategories and further enhances network efficiency.

3.2. Intelligent Session Mapping Algorithm

The algorithm uses the knapsack algorithm to intelligently select voice sessions based on their rating factors and network capacities. It iterates through the knapsack results and identifies the sessions to be included in the optimization based on their impact on the cumulative rating factor. Finally, the algorithm updates the inclusion status of selected sessions and returns the updated list.

Algorithm-2

1. Initialization:
   - Set up variables i and w to represent the index and remaining capacity.
   - Initialize an empty list included_sessions to store selected sessions.
2. Knapsack Algorithm Execution:
   - Execute the knapsack algorithm (knapsack_algorithm) with input sessions, rating factors, and network capacities to obtain knapsack_results.
3. Iterative Selection Loop:
   - Iterate through the knapsack results using a while loop.
   - Check if the current knapsack result is different from the result above in the table.
   - If true, include the current session in included_sessions and update the remaining capacity w by subtracting the session’s weight.
4. Update Session Inclusion:
   - After completing the loop, update the inclusion status of selected sessions in the selected_sessions list based on the indices stored in included_sessions.
5. Return Result:
   - Return the updated selected_sessions list, reflecting the intelligent allocation of sessions based on the knapsack optimization.

The intelligent session mapping algorithm intelligently selects voice sessions to maximize the cumulative rating factor based on their rating factors and network capacities. Thus, it contributes to an efficient and prioritized allocation within the medium access subcategories.

3.3. Optimized Dynamic Voice Session Mapping Algorithm

The optimized dynamic voice session mapping algorithm iterates through AC_VO_3 sessions and computes the real-time rating factors during the computational intervals. The algorithmic optimization is applied during tuning time intervals, where voice sessions are intelligently mapped in either AC_VO_1 or AC_VO_2 based on their rating factor score.

This integration ensures the efficient utilization of network resources and prioritizes voice sessions based on their real-time characteristics [12].

```
Algorithm-3
AC_Voice captures the medium
For each Voice session [i] subject to AC_voice_3
  If computational time == 1
    For each Voice session [i]
      Compute the Rating factor
    End for
    While tuning time == 1
      If Rating factor <= 80
# Apply Algorithmic Optimization for session inclusion in AC_voice_1
 selected_sessions = algorithmic_optimization(sessions, rating_factors, capacity_AC1, selected_sessions)
      Else Rating factor >= 80
# Apply Algorithmic Optimization for session inclusion in AC_voice_2
 selected_sessions = algorithmic_optimization(sessions, rating_factors, capacity_AC2, selected_sessions)
      End if
    End while
    Wait for tuning time == 1
  End if
End for
```

4. Performance Evaluation

The proposed optimized dynamic Medium Access Control (MAC) protocol performance is rigorously evaluated through comprehensive performance assessments using the Network Simulator 2 [6]. The simulation evaluates the protocol's effectiveness in enhancing voice communication quality within MANET's environment compared to the existing EDCA scheme under various network and traffic conditions.

The simulation environment created the network topology carefully crafted to mimic MANET characteristics and included an Access Point (AP) in infrastructure-less mode. The network is configured to handle diverse traffic types, such as background, best effort, and Voice over Internet Protocol (VoIP) calls utilizing the G.711 codec. Each node is assigned a fixed transmission range of 250 meters, a common setting supported by most real-time network interface cards.

All nodes are outfitted with an IEEE 802.11 network interface card functioning at a data rate of 2 Mbps. Random traffic spells are generated between selected source-destination pairs to cope with the dynamic nature of ad hoc networks with multiple voice spell transmissions over the 802.11e network. This comprehensive traffic generation aims to replicate the challenges encountered in achieving Quality of Service for voice communication. The assessment of network performance primarily focuses on two key metrics: Throughput and Packet Delivery Ratio (PDR). For voice traffic, characterized by a Constant Bit Rate (CBR) of 160 bytes' payload and a 40-byte RTP/UDP/IP header overhead for G.711 VoIP calls, throughput is measured. Packet Delivery Ratio is determined as the ratio of voice packets received at the destination's application layer to the total number of voice packets transmitted by the source's application layer. Each simulation extends over a duration of 15 minutes, with sampled data averaged over five runs to ensure robustness and consistency in the results. Figures 1 and 2 illustrate the outcomes.

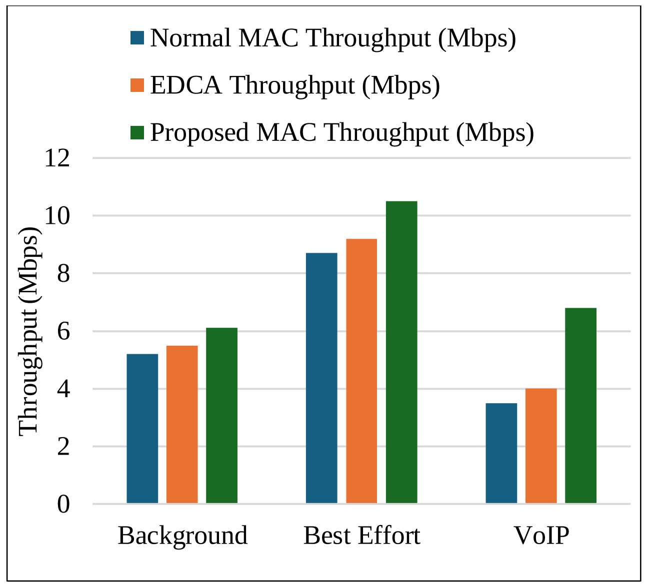

Fig. 1 Throughput comparison between proposed MAC protocol and EDCA under different traffic conditions

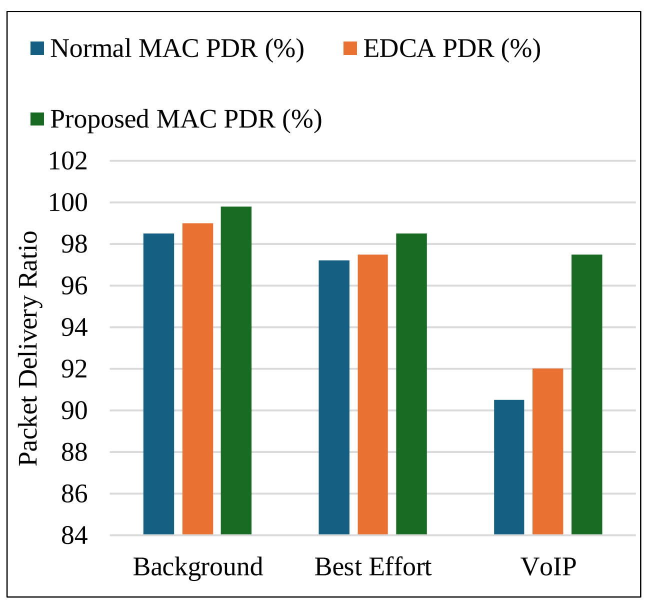

Fig. 2 Packet Delivery Ratio (PDR) comparison between proposed MAC protocol and EDCA under different traffic conditions

Results indicate that the proposed MAC protocol improves performance in both Throughput and Packet Delivery Ratio compared with the dynamic session mapping MAC algorithm under diverse traffic conditions.

The proposed algorithm optimization effectively optimizes the allocation of voice sessions to respective sub-access categories, which reduces packet loss, enhances packet delivery, and minimizes delays. The improvement in results is due to its adaptability to dynamic conditions, ability to prioritize voice sessions based on real-time characteristics, and effective resource allocation optimization.

Thus, compared with the existing dynamic session mapping algorithm, the proposed algorithm significantly enables high-quality real-time communication applications like VoIP within the constraints of MANETs. Thus, the proposed work contributes to enhancing the capabilities of ad MANETs by introducing an optimized and dynamic session mapping MAC protocol. This work aims to advance the overall quality of voice communication in dynamic network environments, particularly in scenarios where traditional infrastructure is challenging to establish.

5. Conclusion

The work proposes an optimized dynamic session mapping algorithm to improve voice communication in the MANET environment. The protocol's innovation lies in its adaptability to dynamic conditions, ability to prioritize voice sessions based on real-time characteristics, and effective resource allocation optimization. The performance of the developed work is computed with Network Simulator 2, and the results show that the proposed algorithm optimization effectively optimizes the allocation of voice sessions to respective sub-access categories, which reduces packet loss, enhances packet delivery, and minimizes delays.

The improvement in results is due to its adaptability to dynamic conditions, ability to prioritize voice sessions based on real-time characteristics, and effective resource allocation optimization. Thus, compared with the existing dynamic session mapping algorithm, the proposed algorithm significantly enables high-quality real-time communication applications like VoIP within the constraints of MANETs. Thus, the proposed work contributes to enhancing the capabilities of ad MANETs by introducing an optimized and dynamic session mapping MAC protocol.

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