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Improvement and Performance Evaluation ofthe Algorithm for Voice over Internet Protocol (VoIP) in Wireless Networks using Artificial Neural Network (ANN)

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Abstract:
One of the most rapidly growing wireless technologies is the so-called voice over internet protocol (VoIP). VoIP does not only find application in wireless networks but also in wired networks. This is majorly because of the fact that VoIP is very much compatible with devices of IEEE 802.11 standard. Besides, its compatibility with the carrier-sense multiple access with collision avoidance (CSMA/CA) makes VoIP a preferred technique for use in wireless communications. This is also because it functions well in 2nd layer of data-link-layer. Among the various algorithms used for VoIP is the distributed coordination function (DCF). But this algorithm suffers from packet collision which occurs when there is a simultaneous transmission of signal without giving attention to other stations' priorities. It is because of this that enhanced distributed channel access (EDCA) is used often to avoid collision as much as possible. However, DCF can still be improved to overcome this obvious setback. In this work, DCF algorithm was improved to get a new algorithm called "distributed coordination function" (N-DCF). This was done by adjusting the competition window of the DCF algorithm repeatedly using Artificial Neural Network technique. The results obtained from N-DCF were compared with DCF in terms of packet loss, jitter, latency and throughput. It was found that for the four metrics, the N-DCF algorithm performed better than the DCF algorithm.
Keywords: *Artificial neural network, Distributed coordination function, Packet loss, VoIP*

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I. Introduction

In today's digital age, Voice over Internet Protocol (VoIP) has become an integral part of our daily communication. It allows us to connect with friends, family, and colleagues across the globe with the convenience of the internet. VoIP services have made long-distance communication more accessible and cost-effective than ever before. While VoIP has significantly transformed the way we communicate, it faces unique challenges in wireless networks. These challenges stem from the inherent characteristics of wireless communication, such as signal interference, limited bandwidth, and variable network conditions.To ensure a seamless VoIP experience in wireless networks, it is crucial to develop efficient algorithms that can adapt to these challenging conditions. The performance of VoIP algorithms in wireless networks is often measured in terms call quality, latency, and path loss. In recent years, there has been a growing interest in leveraging artificial neural networks (ANNs) to enhance the performance of VoIP algorithms. ANNs, a subset of artificial intelligence (AI), have shown remarkable capability in solving complex problems, including those related to network optimization and data analysis.

The success of VoIP in wired networks is well-established, but when it comes to wireless networks, several unique challenges must be overcome. Wireless networks are characterized by signal interference, limited bandwidth, and dynamic network conditions. These factors can lead to packet loss, jitter, and latency, as mentioned previously, significantly degrading the overall call quality. Furthermore, network congestion and varying signal strengths can exacerbate these issues, making it a complex problem ensure a reliable and high-quality VoIP communication.

ANN have gained prominence in addressing complex problems in various domains. ANNs are computational models inspired by the human brain's structure and functioning. They consist of interconnected nodes, or artificial neurons, that process information and learn from data. ANNs are known for their ability to recognize patterns, adapt to changing inputs, and make predictions. These characteristics make ANNs a promising tool for improving VoIP performance in wireless networks.One of the key areas where ANNs can make a difference in VoIP is in the development of adaptive algorithms. Traditional VoIP algorithms often rely on static configurations and parameters, which may not be well-suited for wireless network conditions. ANNs can be trained to adapt in real-time, making them capable of adjusting parameters based on the current network status. This adaptability can help mitigate issues of jitter and packet loss, ultimately leading to better call quality.

Evaluating the performance of VoIP algorithms is essential to ensure that the proposed improvements are effective. Traditionally, metrics such as Mean Opinion Score (MOS), Packet Loss Rate (PLR), and Round-Trip Time (RTT) are used to assess VoIP quality. In the context of using ANNs, new evaluation metrics and methodologies may be required to measure the algorithm's adaptability and real-time performance. The introduction of ANNs into VoIP algorithm evaluation introduces a new layer complexity and opens up avenues for innovative assessment techniques.

This paper aims to provide a comprehensive understanding of the enhancement and evaluation of VoIP algorithms in wireless networks using ANNs. It will explore the challenges faced by VoIP in wireless networks environments, the theoretical foundations of ANNs, and how these networks can be integrated into VoIP algorithms. The algorithms have three main sectional parts: The part that packetizes and depacketizes the signal that is encoded, the part that carries out the function of signal coding and decoding, and the part at the receiver side that makes the delay smooth [9][17][18].

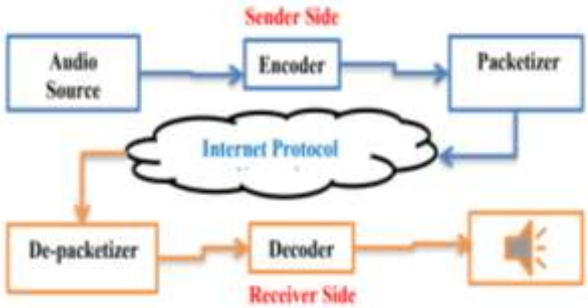

Fig. 1: Process of Communication in VoIP through End-to-End [9]

II. Review of Related Literature

An extensive work on enhancing Voice over Internet Protocol (VoIP) algorithms used in wireless networks was carried out in [1]. Algorithms such as DCF (Distributed Coordination Function) and the EDCA, which stands for Enhanced Distributed Channel Access were considered. To enhance the algorithms, the authors considered the method of parameter modification wherein the parameters of the algorithm were slightly changed to give a new algorithm called M_EDCA, it was mentioned that the new algorithm realized from the modification of the parameters of the EDCA had the best result in terms of performance when compared with EDCA and DCF algorithm.

In the work by [2], the performance of LTE network, particularly for application in voice over internet protocol, was evaluated, with primary focus on the broadcasting mode, which is the downlink. Simulations were performed using simulate, a network simulator. The quality of service (QoS) of voice over internet protocol was measured by adjusting the number of nodes, the distance and the speed from the base station (eNodeB). Packet loss, Jitter, Packet delay, etc. were among the different metrics used for the performance evaluation. It was observed that Long Term Evolution gave better voice over internet protocol quality in the downlink while change in the speed, distance, and the number of users had little effect on the VoIP quality.

The authors in [3] also made a modification of MAC algorithms to be able to give channel access and adequately improve high quality application in VoIP techniques. This enabled the minimization of delays and loss of packets during transmission which results in very poor voice quality during transmission of voice through the network. In [4], the assessment of speech quality in wireless voice over internet protocol communication was conducted. The method adopted was the Deep Belief Network (DBN). Different network scenarios that took into consideration various PLRs, that is, Packet Loss Rates, together with wireless channel models were implemented. It was stated that certain algorithms are not ideal in real voice over internet protocols scenarios. Such algorithms include the algorithm in ITU-T Recommendation P.862.

The work by [5] looked at intelligibility of speech in voice over internet protocol to Public Switched Telephone Network (PSTN) interworking. It was then stated that modification of competition window size can be leveraged on to improve the efficiency and performance of a network. In fact, by imploring algorithms designed for wireless networks in the context of the voice over internet protocol technique and IEEE 802.11 networks, one can enhance or adapt specific algorithms by fine-tuning the competition window size to mitigate or eliminate packet collisions. Reference [6] tested some algorithms and got negative results as regards the quality of packet lost and packet delay in the specific condition of high load. Though reference [1] got positive results as regards collision beams avoidance when a test on competition window and the algorithm was carried out, it was stated that negative results were obtained when there was a selection of inappropriate parameter and values, leading to an increase in packet collisions in the entire network [7].

The authors in [1] reported that adjustments were made to the neighborhood backup algorithm (NBA) used in wireless networks. The modifications involved making the algorithms return between nodes with adjacent nodes at certain intervals, without altering the competition window size, and determining the minimum competition window size in specific nodes [8]. References [10, 11, 12, 13, 14, 15, 16] carried out research work on voice over Internet Protocol (VoIP) in wireless networks using different methods to evaluate, simulate, analyze, review, etc. VoIP.

Artificial Neural Network (ANN)

Artificial neural networks, a component of machine learning, operate based on foundational neural principles. Comprising interconnected artificial neurons resembling their biological counterparts, these networks communicate through weighted connections or edges. Similar to synapses in a biological brain, these connections convey signals in the form of real numbers. Each neuron processes incoming signals, utilizing a non-linear function to determine its output. The weights of neurons and edges are adjustable and evolve during learning, influencing signal strength. Neurons may have a threshold dictating signal transmission.

Training of ANN

In the training phase, neural networks learn by associating known inputs with outcomes, storing these probability-weighted associations internally. Training involves assessing the disparity between the network's output and the target output, known as the error. Guided by a learning rule, the network adjusts its weights based on this error. Successive iterations refine the output, progressively aligning it with the target. Training concludes upon meeting predefined criteria after a sufficient number of adjustments.

III. Materials and Method

Materials

Key materials for this work include the following:
- MATLAB programming software
- Training data
- Personal computer (PC)

Method

The method involves a 6-step approach that includes:
- Data collection and processing
- Training of ANN model
- Evaluation of VoIP algorithm based on the trained ANN model
- Obtaining a new VoIP algorithm by adjusting the competition window (CW) of the already existing algorithm
- Evaluation of the new VoIP algorithm using the trained ANN model
- Checking if the performance metrics are improved.

Fig. 2 shows the system setup that leads to data capturing and eventual training using ANN, while the details of the steps are shown in the flowchart of Fig. 3.

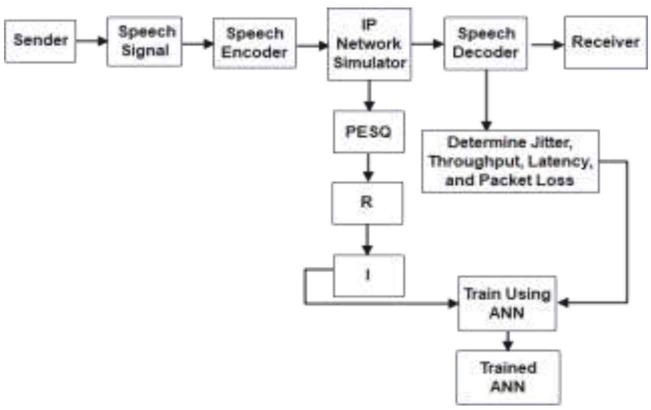

Fig. 2: System Setup for Data Capturing and ANN Training

As shown in Fig. 2, PESQ is the perception evaluation of speech quality which is used for the basis of comparison. R is a parameter used to determine the estimated signal quality, and it has values that lie between 0 and 100. When R is equal to 0 it indicates a very bad signal quality and when it is 100 it indicates a highly impressive signal quality. I is the impairment as a result of jitter, packet loos etc.From the setup, firstly there is an encoding of the speech signal followed by simulation. Thereafter, the output of the simulation is decoded to obtain the faded signal. This faded or degraded signal is compared with the reference input signal by means of the PESQ. At last, the packet loss and other parameters are fed into the ANN Model for training while the value of the impairment I is taken as the target.

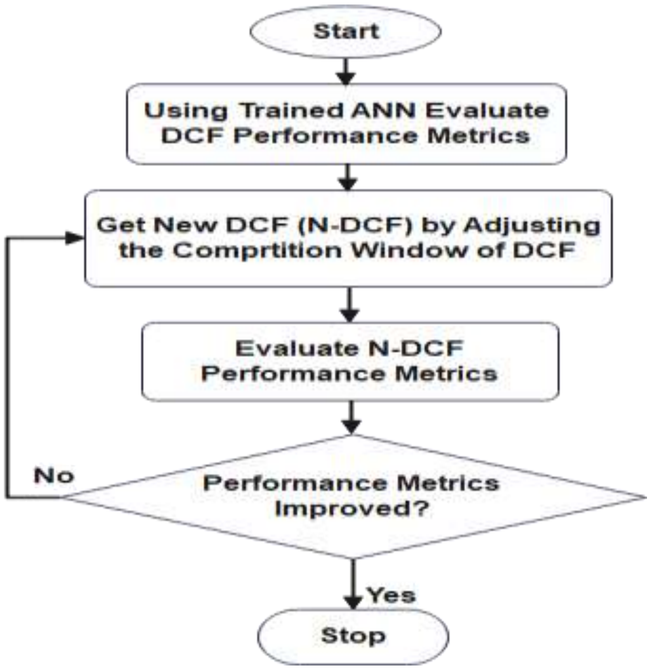

Figure 3: Flowchart that Uses the Trained ANN to Get a New VoIP Algorithm

Competition window (CW) plays a key role in the improvement of VoIP algorithm for better network operation. In this work, as shown in Figure 2, distributed coordination function (DCF) competition window is adjusted repeatedly until an improvement in the performance metrics like latency, packet loss, throughput, and jitter is observed. Matlab programming software is used for the simulation. MATLAB software supports VoIP algorithm, hence suitable for the simulation. Some of the key requirements for the simulation are as summarized in Table 1.

Table 1: Key requirements for the Simulation

Tools	Requirement
Data rate	54Mbps
Simulation period	400s
No. of counts	10, 20, 30, 40, 50

IV. Results and Discussion

To assess the effectiveness of the new algorithm (N-DCF), a performance evaluation was carried out based on such performance metrics as jitter, throughput, latency, and packet loss. The results are compared with those of DCF algorithm. Figs. 4-7 show the results in terms of the four performance metrics: Latency, Jitter, Throughput and Packet loss.

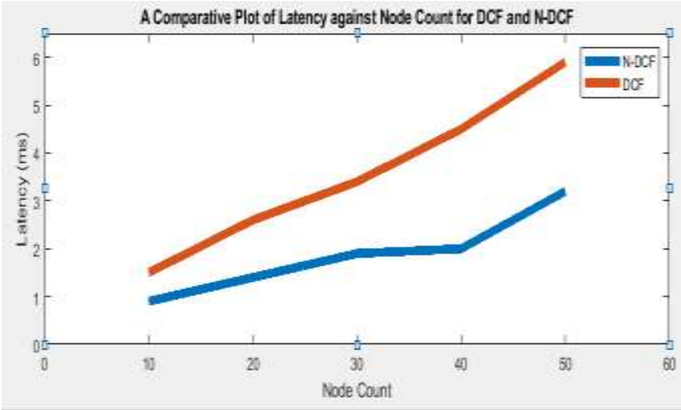

Fig. 4: A Comparison of Latency for Both N-DCF and DCF Algorithms

As can be observed from Fig. 4, the latency for the N-DCF algorithm is lower than the latency for the DCF algorithm. Since a lower latency is indicative of good network performance, it can be seen that with the proposed algorithm (N-DCF), VoIP communication can be enhanced.

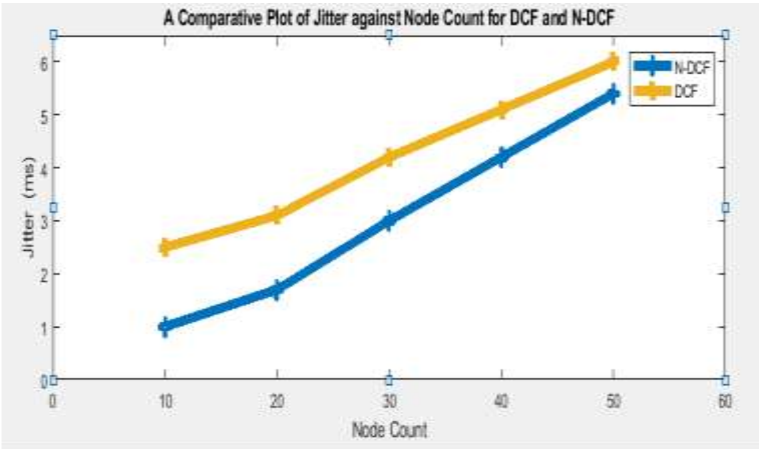

Fig. 5: A Comparison of Jitter for Both N-DCF and DCF Algorithms

Fig. 5 depicts a comparative plot of the jitter values for DCF algorithm and the N-DCF algorithm. It is clear from the plot that the N-DCF algorithm has lower jitter values compared with the jitter values for the DCF algorithm, indicating a better performance of the proposed algorithm in enhancing or improving VoIP communication for wireless networks.

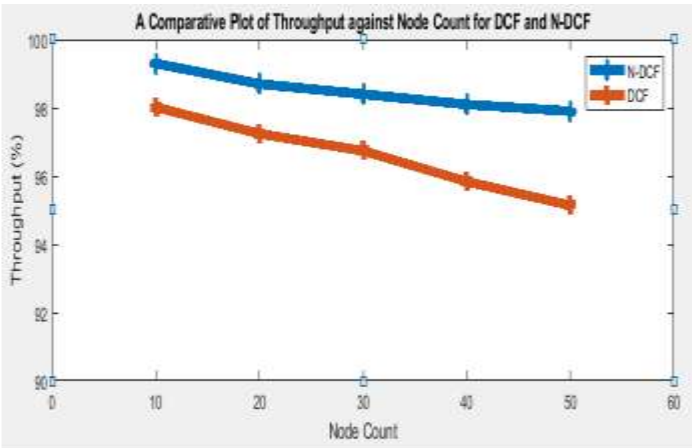

Fig. 6: A Comparison of Throughput for Both N-DCF and DCF Algorithms

In networks, the higher the throughput, the better the network performance. Fig. 6 shows that for the two algorithms (DCF and N-DCF), N-DCF has higher values of the throughput, indicating the superiority of the proposed algorithm (N-DCF) over the DCF algorithm. Hence, it can be said that the proposed algorithm can be relied upon in having VoIP wireless communication improved.

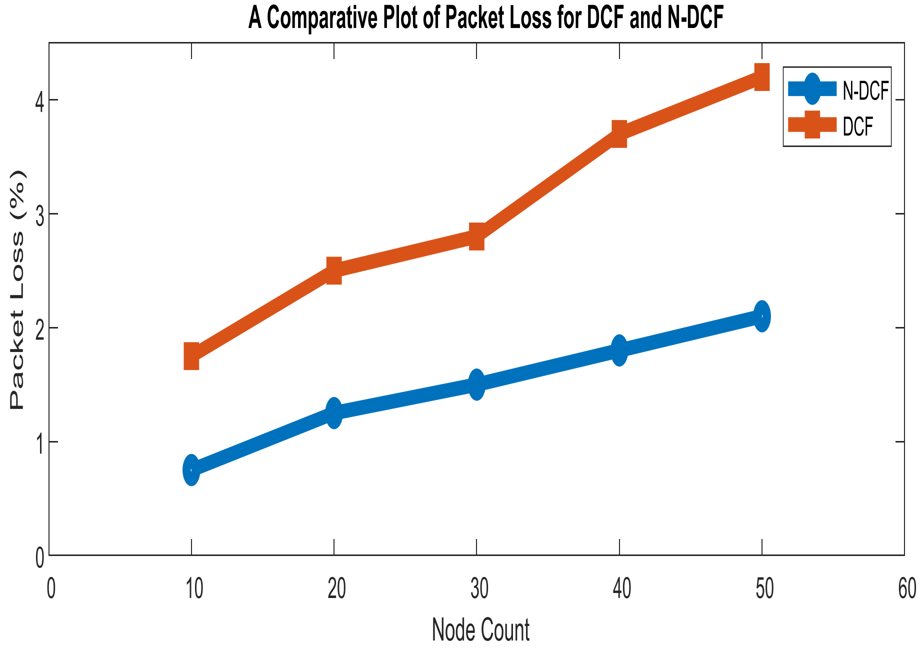

Fig. 7: A Comparison of Packet Loss for Both N-DCF and DCF Algorithms

A higher packet loss in wireless communication is not favorable as it degrades communication. For e better communication, packet loss has to be minimized. Figure 7 shows that of the two algorithms, DCF has higher values of packet loss than the N-DCF algorithm proposed. This indicates a better performance of the proposed algorithm in enhancing or improving VoIP communication for wireless networks

V. Conclusion

An improvement of algorithm for voice over internet protocol (VoIP) has been carried out in this work. There are various algorithms for VoIP but this work only considered an improvement in the distributed coordination function (DCF) algorithm using artificial neural network (ANN). DCF algorithm is known to suffer from packet loss and lower quality of service (QoS) where number of connecting nodes is high and for medium-sized networks. By means of artificial neural network the DCF algorithm was improved, realizing what is called New Distributed Coordination Function (N-DCF) algorithm. The N-DCF and DCF algorithms were compared in terms of performance based on such performance metrics as packet loss, jitter, throughput, and latency. It was observed that in all four-performance metrics, the N-DCF performed better than the DCF, indicating that the new algorithm, (N-DCF), can be used in VoIP for improved wireless communication.

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