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Design of Effective Amplification Signal by Controlling Bandwidth Using Adaptive Learning Technique In Voice Over Internet Protocol

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Abstract:

VoIP refers to the technology that enables the transmission of audio and video in the form of data packets across an IP network, whether it be a private or public one. Voice over Internet Protocol (VOIP) enables many important benefits for both communication service providers and their customers, including reduced costs, enhanced media offerings, mobility, integration, and portability. Despite this, there are a lot of obstacles to VOIP implementation, such as complex architectures, problems with interoperability, problems with handoff management, and security concerns. In particular, the rise in voice over Internet Protocol (VOIP) call transmission is posing a severe threat to more conventional forms of data transmission, such as text messages, as these older methods simply lack up to the task. Some of the difficulties faced by the user is that packet loss, delay, security, Noise, bandwidth overhead and throughput. This research work provides the probable solution effective data transmission by employ to control the bandwidth using the Adaptive call method in clock synchronization.

1. Introduction

In today's 5G networks, Radio Access Terminology is crucial. To provide very dependable, low-latency, and high-speed communication for a variety of uses, 5G network design integrates cutting-edge technologies [1], including network slicing and edge computing [2]. Voice over Internet Protocol (VoIP) is a method of transmitting audio signals from one device to another across the internet. Through the router, these signals may be sent from the source to the destination. A world time can be set up with the help of effective clock synchronization [2].
The exponential growth in the number of people using the Internet is a direct result of the pervasiveness of networks and the ease with which people may access them in today's fast-paced environment [3]. It has a detrimental impact on performance, lowers the offered service quality, and might pose a significant risk to its performance, as mentioned in [4]. This is with added to the fact that traditional IP-based networks that depend on hop-by-hop header inspection for packet forwarding are unable to satisfy the demanding network performance standards (e.g., low latency, high bandwidth, etc.) required by real-time and multimedia applications like voice and video, in addition to ensure quality of service (QoS). Nowadays, several methods like Multi-Protocol Label Switching (MPLS) and others are used [5].

1.1 Wireless Sensor Network

In terms of propagation type, spectrum utilization, and network structure, there are a number of wireless technologies available. The primary benefit of wireless backup is that it eliminates the need for costly and time-consuming connections. Wireless solutions provide faster deployment times and lower costs by requiring just equipment at the point of presence and small networks [6].
It is a set of wireless network protocols that can be established and used without any external infrastructure to track environmental and physical variables like temperature, sound, vibration, and other similar metrics, and then transmit that data to a central location for evaluation. Sinks mediate

communication between users and the network. Once deployed, sensor nodes are free to self-organize in terms of network topology, including cluster head and cluster node selection [7].

1.2 Structure of Wireless Sensor Network

A Sensor network is made up of five main component (figure 1)

a) Sensing Unit
b) Processing unit
c) Transceiver Unit
d) Receiver unit
e) Power Unit'

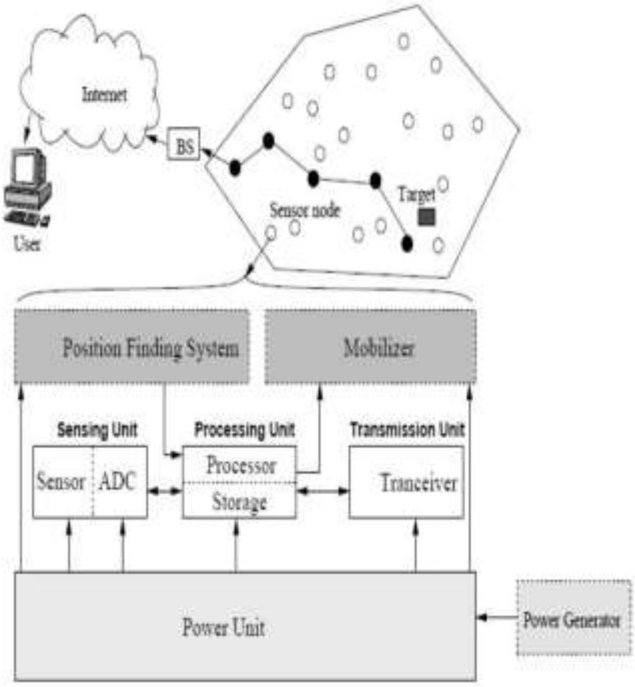

Figure 1. *Wireless Sensor Structure.*

The ADC takes the sensor's analogue impulses and turns them into digital ones, which the processing unit may use. In order to finish the particular nodes, the processing unit works with additional sensor nodes. The node is linked to the network by a transceiver unit, and the sensor nodes are powered by a power unit that is connected to the solar cells. [8].

1.3 Voice Over Internet Protocol

A VoIP service is accessible to anybody with a high-speed internet connection, router, and modem. In a standard VoIP setup, a desk phone and a SIP server—usually a VoIP provider—are the components. With so many more capabilities than analog phone service could ever dream of, it outperforms the old-fashioned landline phone. Your information is safely kept in the cloud since VoIP operates via the internet.

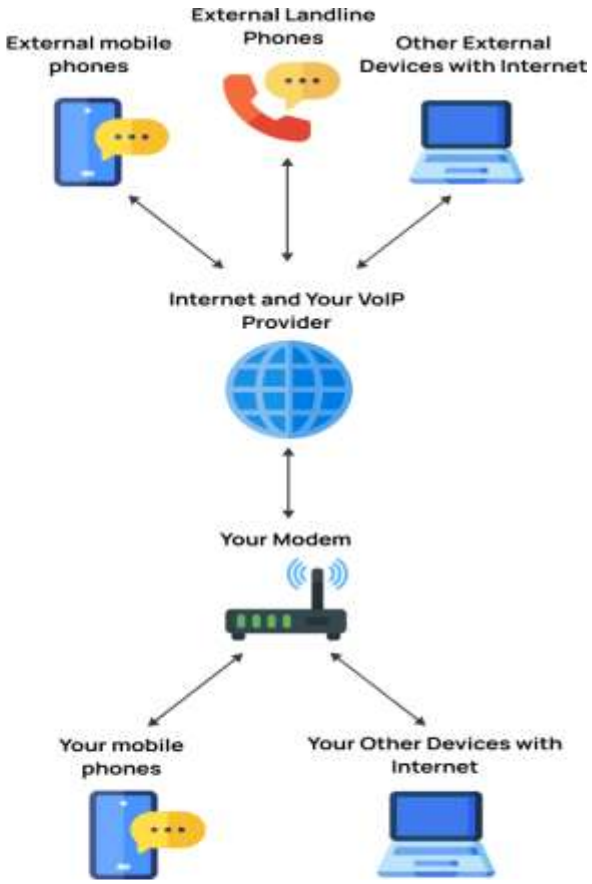

Figure 2. *VOIP call.*

An online dashboard is provided for the management of the VoIP system. phone speaker. One way to avoid using a traditional landline or mobile network altogether is to set up a voice over internet protocol (VoIP) phone system [9]. A voice over Internet Protocol (VoIP) solution uses your broadband connection to digitize analogue speech signals (Figure 2).

To make calls to different phone networks, a VoIP server is used. The call is set up between all participants by the supplier. The receiving end decompresses the digital data into the audible form that your phone or the term "voice over IP" refers to a method of transmitting audio over the internet by digitizing and compressing human speech. A Voice over Internet Protocol (VoIP) service uses the protocols that allow for phone calls to be made over the internet. Phone conversations that bypass traditional landlines by using an existing internet connection [10].

1.4 Clock Synchronization

In sensor networks and other multi-hop ad hoc wireless networks, clock synchronization is an essential issue. It gives various nodes access to the common time frame. In data transfer, it is crucial. Determining when an event occurs is essential in all areas of network management, security, planning, and troubleshooting. The only frame connecting the clocks is time itself. Making sure all the clocks are in sync is vital in the realm of distributed computing, where several systems work together to complete tasks. To facilitate coordinated task execution, efficient data transmission, and seamless communication, clock synchronization is used to synchronize the clocks of computers or nodes.

1.5 Objective

a) To increase the quality of service by employing the effective data transmission between sensor nodes
b) To minimize the delay associated with data transmission during the times of handover and ensure a fair distribution of exchange of data

2. Litreature Survey

Askar Hamdulla and Shijia Liu [11], A time-triggered, globally unified, high-precision time synchronization is the foundation of network deterministic communication, as pointed out in this study. A number of associated activities, including time synchronization, transmission delay, clock drift, and clock synchronization algorithms, will affect the precision of the clock synchronization. It is an enormous technological hurdle to achieve reliable clock synchronization with great accuracy. This article discusses time-triggered Ethernet and also examines the current state of time synchronization technology, a protocol for high-precision clock synchronization, and a literature study on the topic. provides a brief overview of the difficulties associated with clock synchronization before moving on to discuss potential future developments in the field.

In the 2019 study, Muthumalathi, Pankajavalli, and Priya (N. Priya) [12] stated that Clock synchronization is the primary emphasis of packet transfer among sensor nodes. They address the clock synchronization invariances and communication overheads in energy-based techniques by proposing a framework for efficient clock synchronization. Designing and broadcasting sensor network topologies are both aided by clustering techniques. The current protocol, Mob-ZSEP, finds the optimal route by means of a clustering strategy. The suggested clock synchronization methods will make use of the optimal route chosen by the Mob-ZSEP routing protocol. Two new methods, Energy-based Proportional Integral and Least Common Multiple (EPILCM) and Weighted-based Least Common Multiple (WLCM), are suggested in this study. The Energy-Based Proportional Integral (EPI) and the Energy-Based Least Common Multiple (ELCM) are two approaches that are combined in the EPILCM protocol. Reducing energy consumption, synchronization error, propagation delay, clock inaccuracy, and communication overhead, the suggested EPILCM protocol is used for clock synchronization utilizing an energy-based method. Transmission in sensor networks is carried out via the WLCM protocol according to the packet's weight. When the simulation results of the EPILCM and WLCM protocols are compared, the EPILCM protocol is shown to be more effective.

Mrs. G.Saraniya, Dr.C.Yamini [13] proposed Synchronization of time is crucial thing with significant component in wireless sensor network (WSN) to maintain the synchronization using different nodes. Sensor networks have some unique characters which makes difficult for applying the traditional network clock approaches. This paper is about the implementation clock synchronization with HCM model to apply on the LCM and HC factors which will helps to neglect the offset of clock and skewness of the clock rate from sensor nodes. Proposed implementation is used for activating the node to perform synchronization of nodes timely for calculating the Least common multiple of each clock time period in a network. Network will form the cluster group with every node in a network to synchronize the network on time using clock time period. The proposed Cristian algorithms which process the synchronize the time-to-time server which are efficient in model compared with average time synchronization using pair wise messages with the parameters such as accuracy, overhead of communication and computation values.

R.D. Patane [14] and Kaksha Thakare (1) suggested the convergence of personal computers, cellular networks, and the Internet gave rise to the emerging discipline of wireless networking. The shifting paradigm from "anytime, anywhere" to "all the time, everywhere" in terms of access to information is a direct result of the growing interplay between communication and IT. Data security is an important issue due to the growing use of wireless sensor networks (WSNs) in both civilian and military settings. Data transmission quality and security are concerns for wireless sensor networks due to sensor nodes' limited resources, wireless links' broadcast nature, and the difficult deployment environments. Our proposal is a QoS routing method for WSNs that incorporates coding and a selective encryption technique to guarantee data confidentiality and the application-required quality of service while minimizing energy consumption.

Due to the high density of nodes and the restricted communication range, packet forwarding in sensor networks is usually accomplished using multi-hop data transfer. Consequently, routing in WSNs has been considered a major study topic over the last decade. To maximize the use of existing resources and enhance network performance, multipath routing is a commonly used approach in modern wireless sensor networks.

The DEC (Deterministic Energy Efficient Clustering) protocol, which was suggested by Rajesh Chaudhary and Dr. Sonia Vatta [15], outperforms all other current protocols in terms of energy efficiency, is distributive, self-organizing, and is dynamic. DEC's method for letting the sensor network self-organize is simple and effective in lowering computing overhead costs. A wireless sensor network (WSN) is a specific kind of wireless network that gathers data or information from the field via a network of several sensors connected to a single base station. Energy sensitivity of sensors is the main differentiator between WSN and conventional wireless networks. When planning wireless sensor networks, minimizing power consumption must be a top priority.

Importantly, these sensor nodes often drain batteries and shorten the lifespan of networks due to their high energy consumption during communications. Therefore, to increase the network's lifetime, it is crucial to build efficient and energy-aware protocols. This research aims to increase the energy efficiency of wireless sensor networks by describing the DEC (Deterministic Energy Efficient Clustering) protocol.

3. Research Methodlogy

VoIP uses packetized digital voice communication; the phone itself can be either digital or analog. The voice can be encoded and converted to digital form either before or after packetization. Calls can be made via the IP network due to the VoIP gateway. As in the past, local calls can still be made through the telephone provider. It can choose to divide traffic between the IP network and the ALN using least-cost routing algorithms set up in the PBX, or it can utilize the IP network to make all calls between the sites linked to its VoIP gateway. VoIP calls are not limited to phones that are directly connected to the IP network. VoIP calls to ALN-served phones are referred to as "off-net" calls. It is possible to route off-net calls to a VoIP/ALN gateway close to the destination phone via the IP network. An alternate VoIP setup that does not employ a traditional PBX instead makes use of IP phones. Figure 3 illustrates a simplified representation of a voice over IP (VoIP) telephone system that is linked to a WAN. IP phones are linked to a local area network. Local area network (LAN) calls are possible. Digital speech is encoded and decoded by means of codecs that are built into IP phones. The encrypted voice is also packetized and depacketized by the IP phones. The wide area IP network allows for the making of calls between various locations.

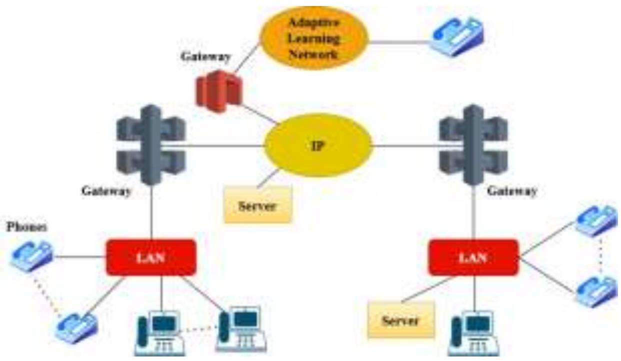

Figure 3. *End to end VOIP*

The primary functions of proxy servers are to register IP phones and to manage call signalling, particularly across different sites. By using VoIP gateways, one can create a connection to the ALN.

3.1 Adaptive Learning Technique

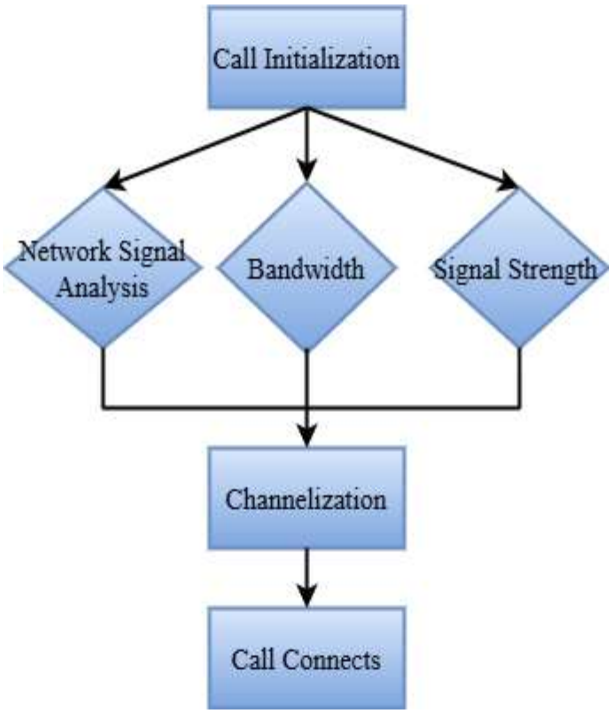

Figure 4. *Adaptive Learning Technique.*

Here we present our ALA, an adaptive learning algorithm that considers the following parameters (figure 4): total delay, codec utilized, packet loss, and burst ratio. A single metric known as the "ALA score" was derived by combining the first three components in the mean opinion score (MOS) with the final factor, latency, which was researched independently and then aggregated with the MOS. While the call is in progress, we utilize it to assess the quality of the discussion. Our method accounts for the delay overhead caused by utilizing multiple Reed-Solomon (RS) codes, which is its major benefit. The ALA score, which is a subjective testing criterion, is used to decide which RS codes to use.

We offer MOS and delay factor ALA levels to find all test cases that need subjective evaluation.

Once the VOIP call is Initialized, it will validate strength of the Bandwidth, Network Signal and Buffer Signal. By Using Adaptive learning algorithm, the sensor node synchronizes with data transmission. While calling, the sender and receiver will connect using WiFi Signal. The frequency of the signal divides depends upon the Spectrum Range.

For each symbol, the relationship between adjacent fully connected layers (denoted as layer p and $\mathrm{p}-1$, where $p \in \{1, \dots, k_{MM} - 1\}$ is as follows: the MM has an input layer, an output layer, and multiple hidden layers, with each hidden layer containing N neurons. The total number of layers in this MM is represented as k_{MM}.

$$cp(i) = Wphk - 1(j) + bk \qquad (1)$$

The loss function takes the form of cross-entropy during offline training; this is a common representation for multi-class classification problems. $Mseq$ is the total number of symbols in the sequence. One way to structure the training is as

$$hk(j) = \sigma(cp(j)) \qquad (2)$$

where N is the number of classes to which a symbol belongs. The anticipated probability $hk(j)$ and the ground truth $p(j)$ are different, and this discrepancy constitutes what we call loss. Each batch comprises a portion of the total $Mseq$ training samples, and the whole training dataset has been divided into batches. Stochastic Gradient Descent (SGD) optimizer with momentum is used to update the network parameters continually. It is much easier than SGD.

$$\min_{Wp,cp\}} Lloss =$$
$$\min_{\{Wk,bk\}} \left(-\frac{1}{Mseq} \sum_{j=1}^{Mseq} \sum_{j=1}^{M} yi(j) \ln(oi(j))\right) \qquad (3)$$

The sequence of received signals after interpolation and zero-mean normalization is represented during equalization as [n^i-l,….,n^i,…,n^i+L], where the vectors n^i,…,n^i+L, $Mseq$ correspond to the symbols received by $Mseq$ in chronological order. For each symbol i, we construct a feature vector x(i) as

x(i)=[n^i-l,….,n^i,…,n^i+K] (4)

The connection is not relevant since we need to develop an algorithm that is codec dependent. As an example, the G.726 16K codec has a maximum MOS of 2.74, which indicates that almost all users are dissatisfied but the G.711 codec has a maximum MOS of 4.41, which means that consumers are very satisfied. The objective is to evaluate the codecs by comparing their performance. Accordingly, using the G.726 16K codec should provide a satisfactory result in the previous example, however using the G711 codec will produce an unacceptable result.

The assigned bandwidth B changes depending on the available bandwidth l, which changes due to factors like packet loss, jitter, and delay in the network. For each given quality of service (QoS), the adaptive learning model can forecast the ideal distribution of bandwidth, denoted as B optimal.

$$B_{AL} = B_{AV} - \Delta\mathrm{B}.\,\mathrm{f}(P_L, J_R, L_R) \qquad (5)$$

B_{AL}- Allocated Bandwidth
B_{AV} -Available Bandwidth
ΔB -Adjustment Step
P_L -Packet Loss
J_R -Jitter Real
L_R-Latency Real

With the goal of preserving constant quality of sound, the adaptive learning algorithm modifies the amplification gain G according to anticipated network environments. Gain optimization is predicted by the learning model ML_{model} given packet loss, jitter, and available bandwidth.

$$G = fML((B_{AV}, J_R, SNR_R) \qquad (6)$$

SNR_R – Real Signal to Noise Ratio
B_{AV} -Available Bandwidth
J_R -Jitter Real
G - Amplification gain

The variable amplification gain G, as shown in this equation, changes according to both observed data in real time and the predictions made by the adaptive learning model. Taking into account both the state of the network and the quality of the signal, the model constantly adjusts the amplification strength to provide optimal speech quality.

To evaluate the quality of a voice, one can utilize the subjective Mean Opinion Score (MOS). To determine how good a call was, it is often used in VoIP systems. Estimations of the MOS score may be made using metrics such as packet loss, jitter, delay, and signal-to-noise ratio (SNR).

$$MOS = \alpha.\left(\frac{B_{AL}}{B_T}\right) + \beta.\left(\frac{SNR_R}{SNR_{Max}}\right) - \gamma.(P_L) \qquad (7)$$

MOS -Mean Opinion Score
B_{AV} -Available Bandwidth
B_T – Total Bandwidth
SNR_R – Real Signal to Noise Ratio
SNR_R -Maximum Signal to Noise Ratio

P_L -Packet Loss
α, β, γ -Weights
To rephrase, to determine whether call quality improvement is necessary, requires a relative relation that allows us to evaluate call quality according to the codec utilized. Since the G711 codec can achieve higher call quality compared to the G726 16K, it might be a potential goal to enhance call quality in the previous example. We separated the peak performance of various codecs under no network impairments into three equal bands to carry used this data to construct the ALA model for the MOS, which depends on the codec used. This enabled us to tackle the problem.
The delay has a noticeable effect on the quality of VoIP conversations. To accommodate our ALA model, that we can easily divided delay ranges according to ITU-T Recommendation G.114 for one-way delay transmission time.

Figure 5. *VOIP in ALN.*

The steps to link the computer to the web are shown in the figure 5. A modem, which the computer communicates to, transforms digital signals into a form that can be sent across the Access Local Network (ALN), which is a local area network. The modem then transmits this data to the ISP, or Internet Service Provider, that acts as the portal to the worldwide web. The user's computer communicates with the enormous network of servers and systems that comprise the internet, and the ISP is responsible for managing the flow of data, directing requests and answers between the two. Websites, apps, and internet services can be accessed by the computer with ease with this configuration.
Maintaining high-quality Voice over Internet Protocol (VoIP) communication is the primary goal of the Bandwidth-Adaptive Signal Amplification (BASA) pseudo-code (Algorithm 1). The code relies on real-time adjustments to both bandwidth and signal amplification. The process starts by configuring the network's capacity and establishing limits for important performance indicators including packet loss, jitter, and delay. While on a VoIP conversation, it keeps monitoring on these

Algorithm 1. *Adaptive Learning-Based QoS Optimization (ALQO)*

```
FUNCTION BandwidthAdaptiveAmplification()
  INITIALIZE Bandwidth B ← B_0
  DEFINE Thresholds:
    PacketLossThreshold P_loss_th
    JitterThreshold J_th
    LatencyThreshold L_th
  SET AdjustmentStep ΔB

  WHILE VoIP_Call_Active:
    // Step 1: Monitor Network
    Measure AvailableBandwidth B_avail
    Measure PacketLoss P_loss
    Measure Jitter J_real
    Measure Latency L_real

    // Step 2: Adjust Bandwidth and Amplification
    IF P_loss > P_loss_th OR J_real > J_th THEN
      Decrease Bandwidth: B ← B - ΔB
      Increase Amplification: Gain G ← G + ΔG
    ELSE IF P_loss < P_loss_th AND J_real < J_th THEN
      Increase Bandwidth: B ← B + ΔB
      Decrease Amplification: Gain G ← G - ΔG
    END IF

    // Step 3: Process Signal
    Amplify Signal with Gain G
    Transmit Signal over Network

    // Step 4: Feedback
    Update Metrics from Network Feedback
  END WHILE
END FUNCTION
```

Algorithm 2. *Adaptive Learning-Based QoS Optimization (ALQO).*

```
FUNCTION AdaptiveLearningQoSOptimization()
  INITIALIZE Model ML_Model
  INITIALIZE Bandwidth B ← B_0
  TRAIN ML_Model with HistoricalData (NetworkMetrics, SignalQuality)

  WHILE VoIP_Call_Active:
    // Step 1: Monitor Network and Input Metrics
    Measure AvailableBandwidth B_avail
    Measure PacketLoss P_loss
    Measure Jitter J_real
    Measure SignalToNoiseRatio SNR_real

    // Step 2: Predict Optimal Parameters
    [OptimalBandwidth, Gain] ← ML_Model(P_loss, J_real, B_avail, SNR_real)

    // Step 3: Adjust Bandwidth and Gain
    SET Bandwidth B ← OptimalBandwidth
    SET Amplification Gain G ← Gain

    // Step 4: Process Signal
    Amplify Signal with Gain G
    Transmit Signal over Allocated Bandwidth B

    // Step 5: Quality Assurance
    Measure QoS_Metrics (e.g., MOS)
    IF QoS_Metrics degrade THEN
      Retrain ML_Model with Updated Data
    END IF
  END WHILE
END FUNCTION
```

metrics to see how the network is doing right now. In the event that things go wrong, like there's a lot of jitter or packet loss, the algorithm will lower the bandwidth to avoid congestion and boost the signal to make up for any possible loss in speech quality. On the contrary, the algorithm boosts call quality by increasing bandwidth and decreases amplification to prevent resource utilizing if the network conditions improve and fall below the set criteria. This adjustment method is designed to function in a feedback loop, allowing the system to adapt dynamically to changes in the network while the call is in progress. For small-scale or stable VoIP systems with relatively constant network circumstances, the BASA algorithm's ease of implementation makes it a great choice. Although it could prove up to the task in very dynamic or large-scale environments, its simplicity guarantees effective bandwidth utilization and stable signal quality. By using machine learning, the Adaptive Learning-Based QoS Optimization (ALQO) pseudo-code (Algorithm 2) takes a more advanced approach to forecasting and optimizing bandwidth allocation and signal amplification. Starting with past data on network metrics including packet loss, jitter, and signal-to-noise ratio (SNR) and user experience metrics like Mean Opinion Score (MOS), the system trains a machine learning model. To ensure high-quality VoIP communication in real-time, the system gathers real-time network measurements and feeds them into the trained model. The model then forecasts the ideal bandwidth and signal amplification gain. The system is able to improve performance proactively according to the current network status by dynamically applying these predictions. On top of that, the algorithm is always checking quality-of-service (QoS) metrics to make sure it's working efficiently. Retaining the model with new data allows it to adjust to evolving network circumstances in the event that performance decreases. The system's ability to learn and develop is assured by this feedback mechanism. In situations where network circumstances are constantly changing, the ALQO algorithm works well for large-scale or complicated VoIP systems like cloud services or international business networks. Modern, dynamic network environments are perfect for its intelligent, data-driven optimization and remarkable adaptability. To fully understand the human perception under a combination of many QoS parameters, a unique closed-network testing approach is used, which incorporates subjective testing of humans. To build a redundancy control algorithm that takes this tradeoff into account, we need to understand it from the perspective of human perception. This set of experiments uses Dummy net to allow participants to score the quality of experience (MOS) of interactive VoIP conversations across a variety of packet loss rate and latency configurations. The Spectrum Range devices may be short orlong that can operate in Single- or two-way Communication mode. The range devices are like Analog radio, Aircraft navigation, Marine radio, Amateur radio, TV broadcasting, Mobile networks, Satellite systems. Once all the Signal Amplified then the Call Connects between Senderand Receiver.

4. Results & Discussions

This section discusses the Analysis and design of control of bandwidth. In this Simulation, Comparing the CallSignal, Buffer Signal, Proportional Signal four repeated call by using new methodology. The transmission takes place using VOIP protocol using Cristian method with Proportional Integral. The Cristian Algorithm helps to Analyze the Signal Transmission. This Comparison calls between the nodes using Adaptive Learning Algorithm. This gives the better results that on the fourth call the bandwidth is controlled and with error-free data transmission. Figure 6,7,8,9 is first to four Call Activated respectively. Figure 10, 11,12,13 are Call 1 to 4 Amplification control.

4.1 Control Bandwidth Signal

Improving the efficiency of data transmission from sensor nodes to the sink while making the most of available bandwidth is the main obstacle. Making optimum use of the available bandwidth is the primary goal of the design.

4.2 Amplification Simulation

Amplification is important to compensate for the loss of strength during its transmission across the channel. It is done at the point in the channel where the strength of the signal become weekend

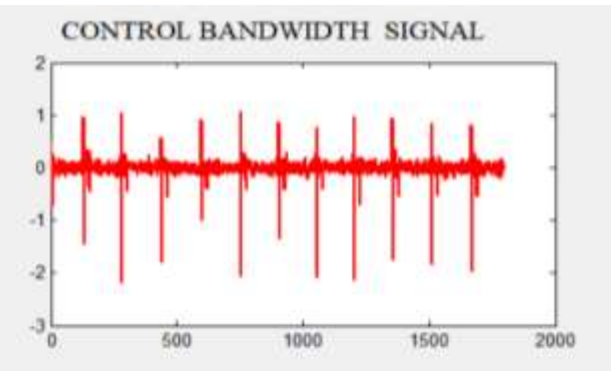

Figure 6. *First Call Activated.*

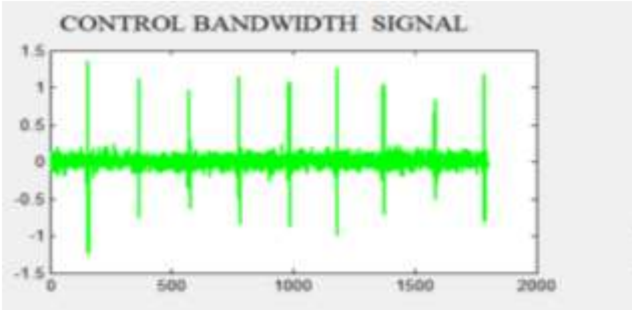

Figure 7. *Second Call Activated.*

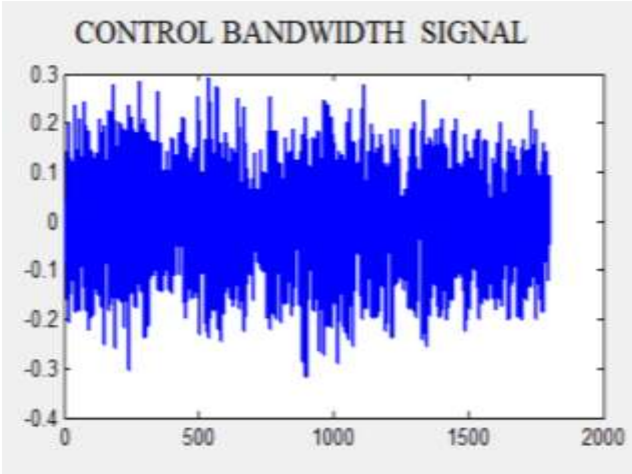

***Figure 8**. Third Call Activated*

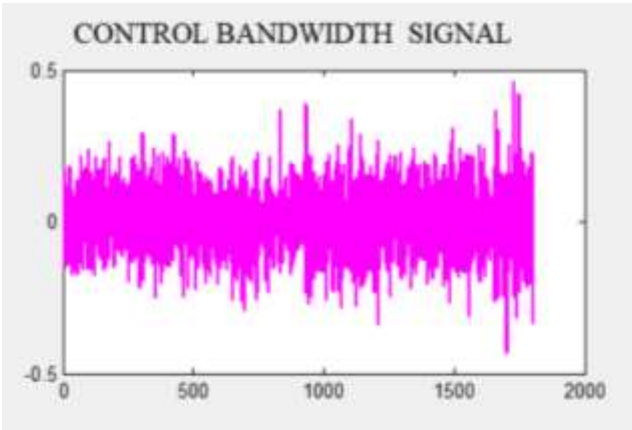

Figure 9. *Four Call Activated*

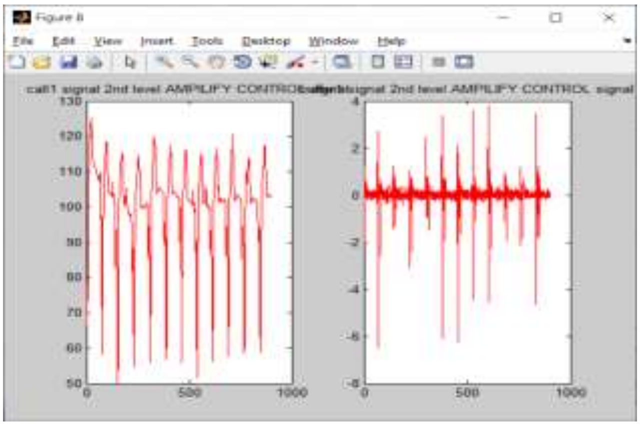

Figure 10. *Call 1 Amplification control.*

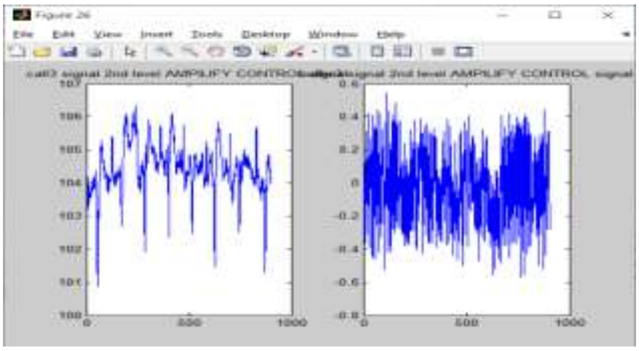

Figure 11. *Call 2 Amplification control.*

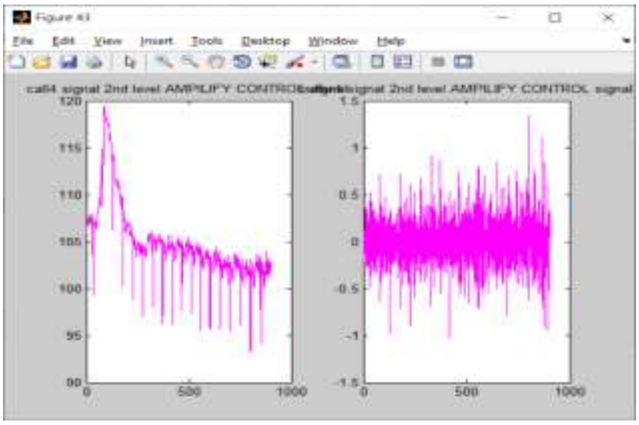

Figure 12. *Call 3 Amplification control.*

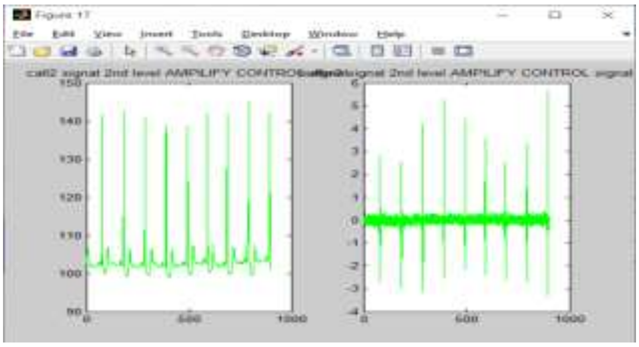

Figure 13. *Call 4 Amplification control.*

4.3. Efficiency and throughput Analysis

By constantly optimising the use of network resources and guaranteeing high-quality voice transmission in VoIP systems, the suggested solution proves to be more efficient (Figure 14 (a) and (b)). This technology dynamically adjusts to changes in the network, including packet loss, jitter, and delay, as opposed to conventional methods that depend on fixed bandwidth allocation and static signal amplification. It keeps the network from being overloaded and avoids resource waste by regulating bandwidth utilization according to real needs. At the same time, the adaptive learning-based signal amplification adjusts the signal's strength so that it remains clear without distorting it or using too much power. Consistently high performance with little resource utilization is ensured by this dual

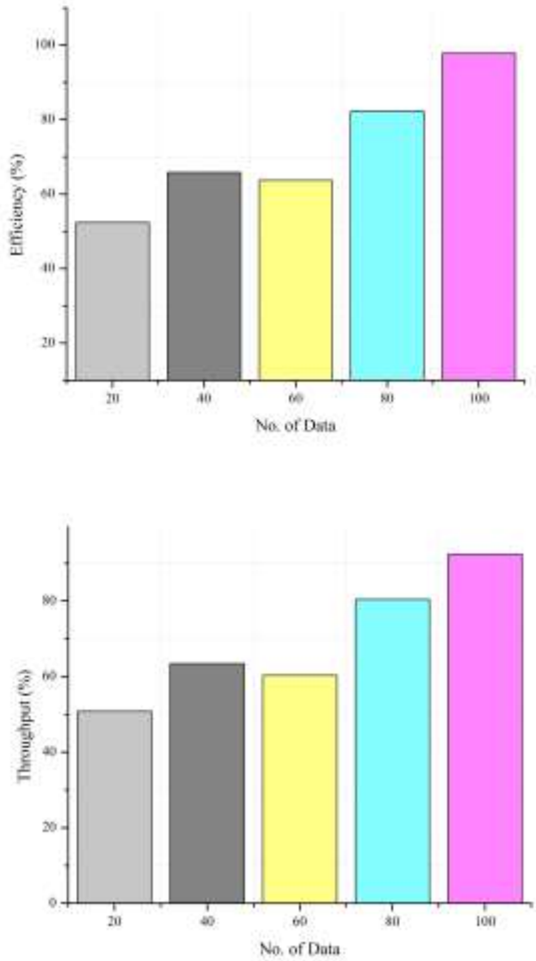

Figure 14. *(a) Efficiency Analysis (b) Throughput Analysis.*

flexibility, which allows the system to function effectively across diverse network environments. If we compare the suggested strategy to static or threshold-based approaches, we discover that it significantly improves resource efficiency.

4.4. Data Transmission Rate and Packet Delivery Ratio Analysis

By combining adaptive learning with dynamic control mechanisms, the suggested solution outperforms conventional methods by increasing the data transmission rate and packet delivery ratio (PDR) in VoIP systems (Figure 15 (a) and (b)). To keep bandwidth high and congestion to a minimum, the system optimizes the transmission rate using real-time bandwidth monitoring and dynamic allocation. To further minimize losses caused by weak signals or congestion, adaptive signal amplification takes into account network characteristics such as signal-to-noise ratio (SNR) and jitter to guarantee strong packet delivery.

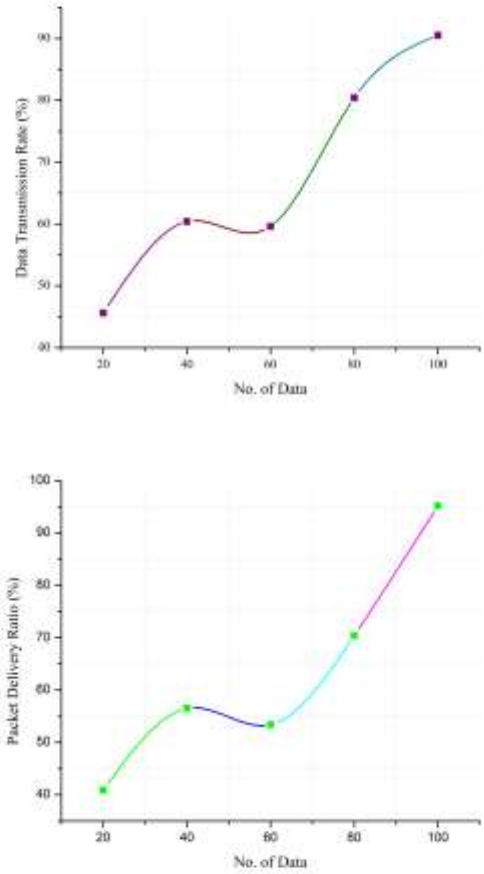

Figure 15. *(a) Data Transmission Rate (b) Packet Delivery Ratio Analysis.*

By anticipatorily modifying bandwidth and amplification parameters according to network variations, predictive modeling further reduces packet loss. When used together, these characteristics provide a consistently high PDR and data transmission rate, surpassing static or threshold-based approaches that cannot adapt to changing network conditions. The Comparison (Table 1 and Figure 16) of signal strength with mode value is used for this simulation. A mode value of 50.03 to 106.01 is the system improvement ratio compared to each adaptive learning. According to the simulation results, the proposed method is to control the bandwidth by evaluating the performance of the call signal and the

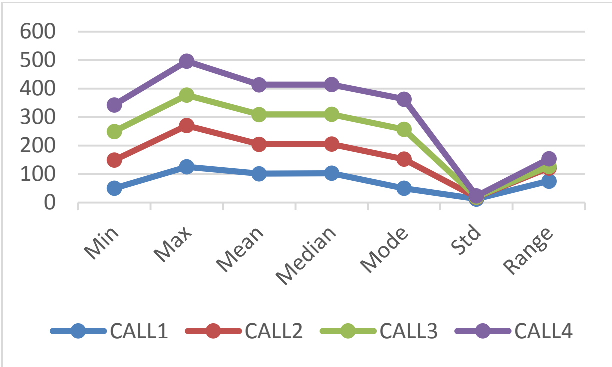

Figure 16. Comparision Analysis

Table 1. *Comparision Analysis.*

Calls	**Min**	**Max**	**Mean**	**Median**	**Mode**	**Std**	**Range**
CALL1	50.03	125.3	101.1	102.8	50.03	12.65	75.28
CALL2	98.83	145.3	103.5	102.4	102.2	5.705	46.43
CALL3	100.9	106.4	104.4	104.4	104.5	0.747	5.488
CALL4	93.3	119.4	104.8	104.4	106.1	4.122	26.12

amplification signal. The correct feedback from the network is obtained by the cristian algorithm with a proportional integral. For every call, the Cristian algorithm calculate the time travel of data transmission. Once Call Signal invoked, Buffer Signal will calculate then Preprocessor signal for both Call Signal and Buffer Signal then two level of error signal for call signal and buffer signal then two level of Amplify control for call signal and buffer signal. In this Experimental Analysis, the skew rate of looping call will calculated.

5. Conclusion

The integration of VOIP into the wireless environment has become the new challenge for the telecommunication world. The limited bandwidth and other constraint present in the wireless environment limit the performance. VOIP is relatively new and emerging area for researcher. Security awareness can take many dimension from different area of IT. However as regarded to VOIP technology, there is relatively little or no research on how organization can have proper measure proio to the implementation of VOIP technology with high level bandwidth. Adaptive Learning is popular method and used in different application recently [16-28].

Author Statements:

- **Ethical approval:** The conducted research is not related to either human or animal use.

- **Conflict of interest:** The authors declare that they have no known competing financial interests or personal relationships that could have appeared to influence the work reported in this paper
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