AN OVERVIEW OF QUALITY OF SERVICE MEASUREMENT AND OPTIMIZATION FOR VOICE OVER INTERNET PROTOCOL IMPLEMENTATION

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ABSTRACT
In this paper, we studied the Voice over IP Quality of Service measurement techniques and Quality of Service implementations. We find out techniques that gives better Quality as well as their weaknesses. We have studied E-Model, Mean Opinion Score (MOS) optimization and other traditional algorithms as well as soft computing methods to find Quality of Service of Voice over IP. A thorough study of getting maximum Quality of service is presented here.

KEY WORDS: *PSQM, E-Model, MOS, Performance Evaluation using RNN, Quality of Service mapping using SOM.*

I. INTRODUCTION

Voice over Inter Protocol (VoIP) enables voice data integration over IP networks, reducing the network transmission cost for IP protocol users. It is due to the shared nature of Internet network. Charlie C. Chen & Sandra Vannoy [1] described it as a global learning tool and studied its perspectives. But it is difficult to guarantee the quality of speeches. The Quality of Service (QoS) depends upon various factors, such as delay, packet loss rate, kind of codec used, etc.
There are numbers of QoS measurement techniques which exists in literature [2][3]. ITU-T recommendation P.862[4] for Perceptual Evaluation of Speech Quality (PESQ) standardizes the voice speech quality access in packet based voice communication. To account for other factors such as the speech quality level and end to end delay in estimating the QoS of VoIP [5] the E-Model [6] has been developed. Assem, Haytham, *et al* [7] developed a testing framework that can provide online estimate of audio and video call quality without requiring either end user involvement or prior availability of network traces. On the other hand, these methods only evaluate QoS values; and it is difficult to correlate the QoS labels of all affecting factors. INTERMON technology [8] is used for automated QoS measurement and analysis for VoIP applications. In a different angle Vadivu, A. Senthil, *et al* [9] have done a performance analysis via non Markovian loss system with preemptive priority. Due to their wide uses, PESQ, E-model, MOS and INTERMON are discussed in this paper against measurement techniques.
Whereas for optimization, E-model Optimization Algorithm [10], Source and channel coding algorithm [11], QoS provisioning system [12], use of Self Organizing Neural Network [13] and use of Random Neural Network [14] are included.
The paper is organized in five sections. First section is the introduction part. In second section we have studied different QoS measurement techniques present for VoIP. Next, we have studied all types of QoS implementations. In third section the traditional techniques and in fourth section the technique which uses Neural Networks. The last section deals with the best techniques and their pros and cons.

II. QOS MEASUREMENT TECHNIQUES FOR VOIP

VoIP refers to the integration of telephone service with IP bases applications. It consists of a set of facilities and protocols for managing the transmission of voice packets using IP. IP networks are not very suitable for transporting real time traffic due to its connectionless nature. So, QoS plays an important role for VoIP. Here we discuss various QoS measurement techniques for VoIP.

2.1 PESQ (Perceptual Evaluation of Speech Quality)

It is an industry standard voice quality testing method widely adopted by telephone manufacturers. It is standardized as ITU-T recommendations P.862 [6].
In this the voice quality test algorithm is divided into two categories:

1. Full Reference (FR) Algorithm:
   It has access to and makes use of original reference signal for a comparison. It can compare each sample of the reference signal to each corresponding sample of the degraded signal. It provides the highest accuracy.
2. No Reference (NR) Algorithm:
   It uses the degraded signal for the quality estimation and has no information about the original reference signal. NR algorithms have low accuracy estimates only.

2.1.1 Limitations of PESQ

The FR algorithm can only be applied to dedicated tests in live networks. NR algorithm does not analyze the decoded audio signal but works on analysis of digital stream. Consequently measurements are limited to transport stream analysis. In PESQ, end to end delay for measuring the QoS is of primary importance, thereby rendering it insufficient for VoIP.

2.2 Mean Opinion Score

The traditional measurement for voice quality in telecommunications is the Mean Opinion Score (MOS). MOS was a subjective measurement where listeners in a “quite room” scored call quality as they perceived it as per ITU-T P.800 [15]. The MOS is the arithmetic mean of all the individual scores and can range from 1(worst) to 5 (best) (table 1).

Table 1: Measurement of Mean Opinion Score

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very Annoying

2.2.1 Constraints of Mean opinion Score

MOS are usually obtained by subjective rating stimuli with respect to a criterion like inter or intra media qualities in a presentation. Subjects express their judgments of media qualities according to a given scale. Finally, the scores are averaged across subjects to obtain the final MOS.
Hendrik Knoche *et al* [16], in their paper, find out some constraints where a MOS – based approach may fail to provide significant differentiation for adaptation or, even worse. MOS tend to exhibit some or all of the following properties that could corrupt results:

1. Insensitivity to effects of unconsciousness;
2. Blurring of relevant details;
3. Ignorance of subjects’ perspectives and introspective positioning.

2.3 The E-Model

The E-Model [6, 17, 18] is a computational model developed by the ETSI (European Telecommunication Standards Institute) and standardized by the ITU (International Telecommunication Union) and ITA (Telecommunication Industries Association) that uses voice and network transmission parameters to predict voice quality.
In this model, the parameter that represents voice quality, R is defined [11, 12] as:

$$R = R_0 - l_s - l_d - l_e + A \quad (1)$$

Where,

R : Overall network quality rating (ranges between 0 and 100)
$\mathbf{R_0}$: Signal to noise ratio
$\mathbf{l_s}$: Impairments simultaneous to voice signal transmission
$\mathbf{l_d}$: Impairments delayed after voice signal transmission
$\mathbf{l_e}$: Effects of equipment (codec's)
A : Advantage factor (attempts to account for caller expectations)

2.3.1 R factor for estimating voice quality

While a network is still being conceived, a network planner can use the E Model to estimate its likely quality. The engineer gathers input information from reference tables, enters it into the E Model, and calculates the resulting Transmission Quality Rating (R factor). Table 2 shows the R factor values for interpretation.

Table 2: R factor values for interpretation

User Satisfaction	R Factor Value Range
Very Satisfied	90-100
Satisfied	80-89
Some Users Dissatisfied	70-79
Many Users Dissatisfied	60-69
Nearly all Users Dissatisfied	50-59

2.3.2 Limitations

a. Very stringent environments are required
b. The process cannot be automated
c. They are very costly and time consuming, which makes them unsuitable to be frequently repeated.

2.3.3 Accuracy of the E-Model

The E-Model is designed to provide estimated network quality and has shown to be reasonably accurate for this purpose. It has not been accepted as a valid measurement tool for live networks.

Against ITU recommendations, the E-Model is being marketed to the industry as a live voice quality measurement tool. The ITU-T G.107[6] recommendation states at the beginning of the document that "*Such estimates are only made for transmission planning purposes and not for actual customer opinion prediction (for which there is no agreed-upon model recommended by the ITU-T)*".

2.4 Automated analysis of QoS parameters using INTERMON

INTERMON technology [8] is designed to support actual challenges for automated QoS measurement and analysis for VoIP applications.

It considers the following approaches to monitor and access network QoS for VoIP in inter-domain environment.

- Monitoring of the connection characteristics for VoIP applications based on active QoS measurement of emulated VoIP traffic.
- Detection of "VoIP impairment" delay and packet loss patterns for network connections characterizing the impact of the network delay and packet loss on the quality of VoIP using E-model.
- Detection of delay patterns of network connection, which can be used to support "heuristics" for playback delay adjustment parameters or algorithms at the receiver of the VoIP applications.
- Discovering the impact of inter-domain routing and topology on the network QoS impacting the VoIP quality.

The figure 1 shows the integration of VoIP QoS pattern analyzer in the INTERMON toolkit.

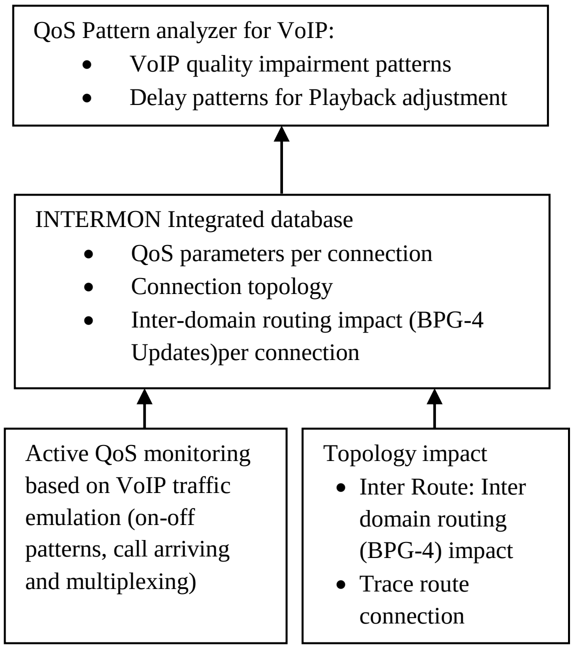

Figure 01: INTERMON approach to study QoS of inter domain connections of VoIP **applications**

2.4.1 Exclusions

INTERMON excluded detection of QoS delay and packet loss caused in the inter domain environment, such as inter domain route change, route flapping, traffic load, etc.

III. TRADITIONAL TECHNIQUES TO IMPROVE QUALITY OF VOIP

There are various factors which affect the Quality of VoIP. Perceived voice quality is an important metric in VoIP applications. The quality is affected by delay, jitter, packet loss …etc. L Sun and E Ifeaher [19] discussed a minimum overall impairment as a criterion for buffer optimization. Traffic is better characterized by Weibull distribution than an Exponential distribution. M J Karam and F A Tobagi[20] in their paper discussed these considering Internet. Z Li, L Zhang, and D Xu [21] also studied about network delay and delay jitter and its effect on VoIP. Voice Quality prediction [22], the QoS management [23], QoS of user experience [24], perceptual QoS assessment [25], Loss recovery techniques [26] of VoIP had been studied earlier. W Jiang, K Koguchi and H Schulzrinne [27] studied the quality and performance of VoIP end points. They studied the aspects of mouth to ear delay, clock skew and silence suppression.
The advent of soft computing techniques based optimization necessitated rethinking about the QoS. Here, we summarize some of the traditional optimization methods and discussed some more methods using the soft computing techniques.

3.1 The E-model Optimization Algorithm

The goal of the optimization algorithm [10] is to place as many calls over a VoIP link as possible without the quality of voice degrading past a minimum quality level. Stated classically, using E-Model quality measures:
Maximize: Number of calls over link
Subject to: R (coding scheme, loss, delay, link bandwidth) >= 70
The optimization problem is set up as follows.
- The *Set* of system configurations is defined.
- The parameters are calculated. This includes all E-Model parameters with fixed inputs and variable inputs based on the combinations in the *Set*.
- Establish the objective, which is to maximize the number of calls on a link.

• Set the first constraint, (R>=70). Set the second constraint (Sum of *Portion = 1.0).*

This yields the following AMPL [28] optimization algorithm.

Set*: CODE;*
Parameters*: T{CODE}, E-Model Parameters,*
Ie{CODE}, MTU {CODE}, Fixed Data Delay, Calculation of E-Model Parameters,
Variables: *Portion{CODE}, Code_Feas {CODE} binary*
Objective:
Maximize Calls: sum {i in CODE}:
*(Code_Feas[i]*portion[i]*LinkBW*util/Rate[i])*
Subject to*:*
Minimum R {i in CODE} : Ro – (Id[i] + Is[i] +
*Ie[i])*Code_Feas[i] + A >= 70;*
Subject to*:*
Total Code: sum {i in CODE}portion [i] = 1;

This E-Model Optimization algorithm presented by M.T.Gardner *et al*[5] where three cases were considered to verify and validate.

Case 1. Find the optimal voice coder given link bandwidth, packet loss level and link utilization.

Case 2. Find the optimal voice coder and optimal packet loss level given link bandwidth and link utilization.

Case 3. Find the optimal voice coder and the optimal link utilization level given link bandwidth and packet loss level.

The optimization algorithm worked correctly. All three programs maximized the total number of calls on the sample voice IP network while maintaining a voice quality level of R=70.

Assem, Haytham, *et al* [29] used a simplify version of E-model to simplify the calculations and focus on the most important factors required for monitoring the call quality.

3.2 Optimization of Source and Channel Coding

In this proposed optimization scheme [11] the constraints are the limited bandwidth and observed Round Trip Time (RTT). The objective is to optimize the expected speech quality using the E-Model. The problem can be formulated as follows.

Given B as the maximum bandwidth for the application, we assume that there are n_c new frames to be transmitted and n_r old frames needed to be retransmitted in the sender queue. The Adaptive Multi Rate (AMR) mode is denoted as m_i for the new frames (i=0..n_c) and mr_i for the old frames in the retransmission queue (i=0.. n_r). In addition to the eight AMR modes (0..7), the mode 8 indicates no retransmission. The bandwidth requirement for each frame is denoted as b_i (new frames) and br_i (old frames) and the Forward Error Correcting (FEC) mode as f_i and fr_i, respectively. Based on the prevailing packet loss rate and the FEC mode chosen (see Table 3).

Table 3: Redundancy overhead and residual packet loss rate (PLR) in different FEC modes

FEC mode	Redundancy	Residual PLR
0	0%	P
1	25%	$P(1-(1-p)^4)$
2	50%	$P(1-(1-p)^2)$
3	100%	P^2

The packet loss probability can be estimated for each frame and marked it with p_i (new frames) and pr_i (old frames). In addition, the expected quality is denoted with Q_i and Qr_i. The quality estimate can be determined using the equation 2.

$$R = 93.2 - I_d - I_e \qquad (2)$$

Where I_d can be solved when the end-to-end delay is known, and I_e, can be solved when the frame loss probability p and AMR mode m are known. If the AMR mode 8 is chosen for a retransmitted frame, quality Q is zero. Because the impact of delay is considered constant, the function for resolving the quality Q can be simplified in the form of equation (3) by removing Id from equation (2):

$$Q(m, f) = 93.2 - I_e \quad (3)$$
$$\max Q = \sum_{i=0}^{n_c} Q(m_i, f_i) + \sum_{i=0}^{n_r} Q(mr_i, fr_i) \quad (4)$$
$$\sum_{i=0}^{n_c} b_i + \sum_{i=0}^{n_r} br_i < B \quad (5)$$

The constraint problem can be formulated in the form of equation (4) that is a subject to the condition (5).
The optimization algorithm can be summarized as follows:

1) Use *Channel Condition Estimator* to estimate the packet loss rate *p* according to the feedback.
2) If there are packets in the *Retransmission Queue*, use the estimated RTT and known pre-buffering delay to evaluate whether these packets can meet the deadline at the receiver side. If the packet is out-of-date, delete it from the *Retransmission Queue* directly.
3) Exhaust all the possible combinations of *mi*, *fi* and *mri*. Use *fi* to compute *pi* as shown in Table 3. Solve *Ie* for each packet using the parameters *mi*, *fi*, *mri* and packet loss rate *p*.
4) Calculate *Qi* and *Qri* using Equation.
5) Sum all of *Qi* and *Qri*, resulting in total quality *Q*. Choose the combination with the maximum value for *Q*.
6) Delete the frames that has been retransmitted and the packets with *Id* higher than 80 from the *Retransmission Queue*.

The results from practical experiments show that this scheme achieves better speech quality compared to a non adaptive streaming system which uses the best-performing combination of speech coding and FEC parameters.

3.3 QoS provisioning System for VoIP

In traditional telephony, there is a call admission control (CAC) mechanism [12]. i.e. when the number of call attempts exceeds the capacity of links, the request for setting up new calls will be rejected while all calls in progress continue unaffected. Several CAC mechanisms such as Site-Utilization-Based CAC (SU-CAC) and the Link-Utilization-Based CAC (LU_CAC) have been used in the current VoIP systems. However, none of these can really provide QoS guarantee to VoIP networks.
The QoS provisioning system [12] can be integrated seamlessly into the existing commercial VoIP systems to overcome their weaknesses in offering QoS guarantee. QoS provisioning system supports two types of QoS guarantees: Deterministic services and statistical services.
Deterministic support applications that have stringent performance requirements for a service without delay bound violations. While they provide a very simple service model to the application, deterministic services, by their very nature, tend to heavily over-commit resources because they reserve resources according to a worst-case scenario. Statistical services, on the other hand, exploit stochastic properties of traffic flows. They allow a predefined portion of packets to miss their deadlines, provide probabilistic performance guarantees and, therefore, significantly increase the efficiency of network usage by allowing increased statistical multiplexing of the underlying network resources.

3.3.1 Constraints of QoS provisioning system for VoIP

The main pitfall of such type QoS provisioning system is that the system has to control or knowledge on the dynamics of the network, particularly on the traffic, in order to provide QoS guarantee. The practicability of a QoS provisioning system relies on the degree of the dependency on such knowledge and control.

IV. USE OF NEURAL NETWORK TECHNIQUES IN QOS IMPLEMENTATION IN VOIP

Learning is one of the capabilities that make artificial neural network (ANN) a favorable approach for prediction and QoS implementation algorithms [30, 31, 32, 33, 34, 35]. Not only ANN, but other soft computing based techniques are used for prediction algorithms due to their non linear operation and flexibility.

4.1 QoS mapping using Self-Organizing Neural Network

A Self-organization based mapping model is an effective tool [13, 36, 37] that can clarify the relative relationships in high-dimensional input data. Based on this method, nonlinear statistical relationships in high-dimensional data can be converted into a two-dimensional space, while preserving the metric and topological relationships of the input data. As a result, this mapping model can be used to evaluate and categorize the relative relationships of high-dimensional input data.
The concept of QoS evaluation of VoIP network communication using Self Organizing Maps (SOM) [13] is shown in the figure 2.

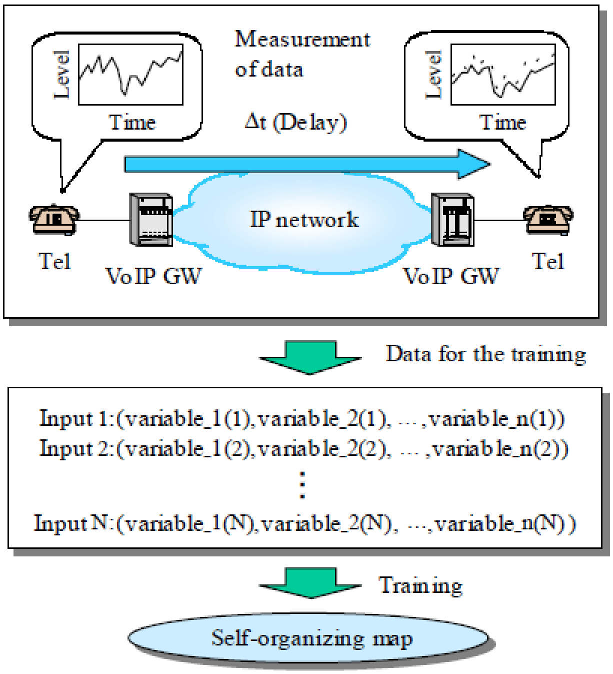

Figure 02: Concept of QoS evaluation procedure for VoIP communication

Here the input data for training the self-organizing map can be given by the target VoIP communication condition. In this training two categories of variables are input into the self-organizing map: one is the QoS related parameters and the other is the system identifiers that indicate the basic system performance level and properties. Here, in terms of the QoS related input parameters to the self-organization map, three parameters PSQM, end-to-end delay, and the packet loss rate were measured in a real environment, and they were used to evaluate the total QoS level among other conditions.
Learning is one of the capabilities that make artificial neural networks and neuro fuzzy systems a favorable approach for time traffic prediction. Supervised neural models, as Multilayer Perceptron (MLP), and Neuro-Fuzzy, as Evolving Fuzzy Neural Network (EFUNN) [34], have demonstrated their capability for solving supervised time prediction problems in many applications. We have applied EFUNN in our work because of the good results obtained in time-series prediction, and because it will also make possible the extraction of underlying classification rules, for future network troubles' resolution.
On the other hand, the use of SOM [13] for the visualization of the status of the communication network is common. The advantage of the SOM capability is utilized for reducing the data dimensionality while retaining as much information as possible. As it is well known the SOM can be viewed as a non linear extension of principal component analysis (PCA). The SOM is used to cluster the network statistics, and to visualize in a 2D diagram (map) in real time when the network has overbooked resources, or when the network has QoS problems.

First, the SOM visualization allows to find a map region where the admission is possible, and to visualize the network status. The objective is to represent in real time the status of the communication network: the SOM provides a straightforward visualization of the current network resources, and allows determining when the network can admit new incoming calls or when it is not possible. Then, the second neural network for traffic prediction forecasts the quality of the new clients for call admission of new incoming calls in order to make an automatic decision system.
MLP and EFUNN[34] were implemented by using a library called Waikato Environment for Knowledge Analysis (WEKA). This is a comprehensive suite of Java class libraries that implements many state-of-the-art machine learning and data mining algorithms. Further, the Data Bionic ESOM tools library for the SOM is also used.
This scenario consists of a group of clients with restricted IP links, and no application or background traffic have been defined, so only VoIP traffic is simulated. The number of calls is increased during simulation; two possibilities are defined for new incoming calls, with QoS or without QoS. The Prediction Resource Manager predicts and visualizes the network status in order to allow or reject new incoming calls.
As criterion for the SOM development the minimum square error (MSE) for the training data is used. The best MSE is achieved with a specified set (typical 80 X 50) of neurons map (related to the two first principal data components), after a fixed training cycles (typical 200).

4.2 Performance evaluation using Random Neural Network

In performance evaluation using Random Neural Network (RNN) [14, 30, 39, 40, 41, 42], RNN is an open Markovian queuing network with positive and negative customers. The server rate at neuron *i* is denoted by r_i and, after leaving neuron (queue) *i*, a customer leaves the network with probability d_i, goes to queue *j* as a positive customer with probability p^+_{ij} and as a negative customer with probability p^-_{ij} .Customers arrive from outside at neuron *i* as positive ones, according to a Poisson process with rate λ_i^+ . Here in this RNN, no negative customers arrive from the environment.
In order to train the neural networks, a database is developed for subjective test results (MOS scores) for different speech samples transmitted under different conditions. The distorted samples are generated using the Robust Audio Tool (RAT) over a LAN in which losses are generated according to the loss model. Using LAN as a context as a test bed to generate the distorted samples allowed us to control precisely the network parameters, which would have been much harder, or even impossible, in a larger network. It has many encoding options, and it is based in the M-bus architecture.
The network parameters that are used are the loss rate (LR) and the mean size of loss bursts Mean Burst Size (MBS). The values used are for the LR (2,7,10 and 15%) and three values for MBS (1,1.7 and 2.5 packets). For encoding the following parameters are considered:

- *Codec*
- *Redundancy*
- *Redundancy offset*
- *Packetization interval(PI)*

A set of original samples are used to generate a number of four sequence groups, one for each of the configurations considered. The original sequences are sampled at 8 kHz (16b mono) and their contents are unrelated. Half of them are male voices and the others are female. The four samples in each group are chosen randomly between the original ones, and then transmitted with RAT over the test network in order to affect their quality as it would be affected by normal usage of such a tool. Finally, a few numbers hidden reference groups are also added to the test, in order to help detect subjects who could not conduct proper evaluations, and to add dynamism to the test's scale.
The results obtained are screened to eliminate subjects that produce erroneous values, and so from the given subjects that originally performed the test, maximum results were used to calculate the MOS, as some are rejected during the screening process. The screening method used is β_2 test, in accordance to the ITU recommendations.
This method aims to overcome the limitations of the available quality measuring techniques in the literature, and it presents several advantages over them:

1. The results obtained correlate well with human perception.
2. It is not computationally intensive.

3. It can be used in real time applications.
4. Some parameters that cannot be easily taken into account by the traditional methods are easy to consider using our approach.

V. CONCLUSION

In this paper we reviewed the topic of speech quality assessment for VoIP QoS in depth. We summarized the various measuring standards available to measure Quality of service of VoIP, reviewed the PSQM [6] for voice quality testing, MOS [15], The E-Model [6, 17, 18] as well as INTERMON [8]. We discussed the traditional approaches used to improve the quality of VoIP which includes the E-Model optimization algorithm [10], source and channel coding optimization algorithm [11], QoS provisioning system [12], their way of measuring and the problems faced by them also reviewed. As many numbers of entities are involved in quality of services analysis, it is not easy to find out an optimal way to tune them by using the traditional methods. The other aspect is to use the neural network or fuzzy techniques [30, 31, 32] to optimize. Here, we have discussed the various neural network techniques [33, 34, 35] used for Quality of service up to now. Among them using Self Organizing Neural Network (SOM) [13, 36, 37], using Random Neural Network (RNN) [40, 41, 42] is important.
The SOM provides a straight forward visualization of the current network resources. So we can determine when the network can admit new incoming calls and when it is not possible. Using Random Neural Network, the network parameters used were loss rate, Mean Burst size and for encoding codec, redundancy, redundancy offset and Packetization interval is used. We conclude that the use of neural network for QoS of VoIP is more advantageous. Here, we can find out over all network traffic position as well as the results can be correlated with human perception. Again it is not intensive in computation. Some parameters cannot be taken easily by traditional methods.

VI. FUTURE WORK

RNN can be used as an effective optimizing tool. In recent years RNN’s are used in various fields of research to get optimized results. For different codec’s, protocols, other components which affect the Quality of VoIP can be measured and optimized by using RNN.

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