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Measuring Quality of Service of VoIP Based on Artificial Neural Network Approach

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Abstract --- Voice over Internet Protocol (VoIP) is one of the fastest growing applications over Internet. The quality of service of VoIP can be measured by ITU E-model which is a subjective measure. In this paper, we discuss a novel method using Artificial Neural Network (ANN) to predict quality on real time traffic. It overcomes the drawback of available methods. NN has learning capability, so it can adapt to changes in the network without any human intervention. Using this method, quality of voice can be calculated at receiving end without any reference voice signal.

Key words --- VoIP, Quality of Service (QoS), ANN

I. INTRODUCTION

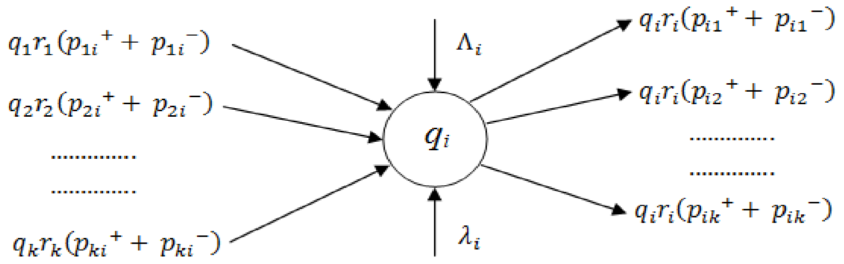
VoIP is a popular application on Internet. The Internet (IP network) is connectionless in nature. It is very difficult to provide Quality of Service (QoS) for real time traffic (Such as VoIP) on such network. In voice communication the perceived voice quality is expressed as Mean Opinion Score (MOS) [01]. It can be measured either by subjective methods which involve human listening and rating it on a scale of 1(poor) to 5 (Excellent) or by objective methods. Subjective methods can be intrusive or non intrusive. Intrusive methods (e.g. PESQ) [02] are more accurate and widely used. But, it is not suitable for live traffic. Because, it needs both the reference voice signal and degraded voice signal to calculate MOS. Non intrusive methods as used by ITU-T E models [03,04] need extensive mathematical calculations. The results from these methods may not always correlate with human perception.

Neural networks has been implemented to predict QoS for VoIP [05,06]. Their training period is time consuming. But a major advantage is, it does not need the reference or degraded signals. It can predict the quality using different network parameters. We have selected the Artificial Neural Network (NN) [07,08] model because it has many features which make it appropriate for our study, discussed later.

The rest of the paper is organized as follows: In section 2 ANN model is presented. Section 3 explains the experiment we carried out. Results are discussed in section 4. Conclusion and future work is discussed in section 5.

II. OVERVIEW OF ANN MODEL

The Artificial Neural Network (ANN)[07,08] is inspired by the spiking behavior of biophysical neurons. When a biological neuron is exited, it transmits signals, called action potentials or spikes along its axon to either excite or inhibit the receiving neurons. In ANN, these signals are represented as excitatory and inhibitory spikes of amplitude +1 and -1 respectively. Each neuron can fire only when its potential is strictly positive. An artificial neuron is shown in the figure 01.



Λ_i	Arrival rate of exogenous excitatory signals
λ_i	Arrival rate of exogenous inhibitory signals
$q_j r_j p_{ji}^+$	Arrival rate of excitatory signals from neuron j
$q_j r_j p_{ji}^-$	Arrival rate of inhibitory signals from neuron j
q_i	Probability that the neuron is excited

Figure 01: An Artificial Neuron

A. Advantages of ANN:

Its standard learning algorithm of ANN has low complexity and strong generalization capacity even for a relatively small training data set. ANN can be easily implemented in both software and hardware since its neurons can be represented by simple counters. The neuron potential is represented as an integer rather than a binary variable resulting in a more detailed system-state description. The stochastic excitatory and inhibitory interactions in the network make it an excellent modeling tool for various interacting entities.

III. SIMULATION AND EXPERIMENT

A. Simulation:

The whole experiment is divided into two parts. Part 1: Training of RNN model using the voice database of degraded signals. Part 2: Test the RNN models to predict MOS.

Here, we used the speech samples available from ITU-T [09]. The network setup for generating degraded voice samples is shown in figure 2. The network setup consists of four client systems (two windows and two Linux systems), one VoIP server and one WANem server [10]. Internet is connected through VoIP server. We have used the SIP based soft-phones (Zopier) [11]. We have installed Asterisk VoIP server [12] on Caldera Linux server. WANem server is installed to emulate the network for different conditions.

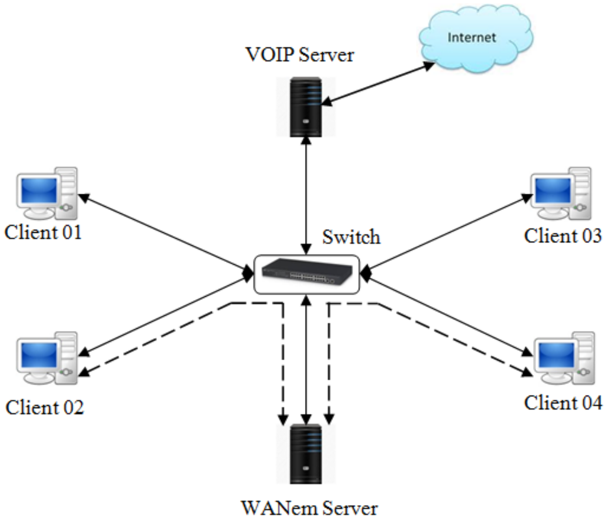


Figure 02: Network setup to get degraded signal

B. Procedure for emulation

Switch on all the client machines, Asterisk server, WANem server. Check for the logical connectivity using ping. After successful logical connection, the network is emulated with WANem emulator. First, we find the IP addresses of source and destination computers. Signals from source computer travel from source to switch then to VoIP server. Signals travel from VoIP server to switch and from switch to destination computer. A logical connection is established between source computer and destination computer through switch, VoIP server then again switch to destination computer. This is the logical path for VoIP signals.

To emulate the network, we have diverted the logical path of VoIP signal packets through WANem server. We have changed the route from source to VoIP server through WANem server. Similarly, we have also changed the route from VoIP server to destination via WANem server. Using WANem server, we have emulated the network packet flow for different values of delay, jitter and packet loss.

C. Experiments:

We switched on all the client, server computers and switch. All computers are checked for the logical connectivity using ping program. Asterisk VoIP server is started. All Zopier phone are started in client machines. Zopier clients are connected to the Asterisk server using the username and password. To connect to another Zopier client, we call using the respective phone number. If the destination computer started ringing then a connection for VoIP can be established or by finding the error numbers we can sort out the problems.

After confirmation of VoIP connection, we diverted the network path through WANem server. In the client system change the network route to the destination computer using route command. Similarly in the destination computer change the route from the source computer. After changing the route, we checked the path using ping software. The result of path from source computer to destination computer came through the WANem server. Using WANem server emulate the network for 1% of packet loss. The test voice sample is sent to destination and recorded. Similarly, emulate the network for 2%, 3% upto 10% loss of packets. All the degraded voice samples for different percentage of loss of packets are recorded. Source sample and the respective degraded recorded signals are fed to PESQ algorithm to find out the MOS of degraded voice sample. We calculated all the respective MOS values of all degraded voice samples. Few samples are taken randomly to train the ANN and others are considered to test the network.

There are two types of architectures of ANN, feed forward and recurrent. Here we have used the feed foreword network with our experiment. The detailed lay-out of the network is shown in the figure 3. The ANN consists of five numbers of inputs (latency, packet loss, jitter, delay, and codec), followed by three nodes of hidden layer and an output neuron.

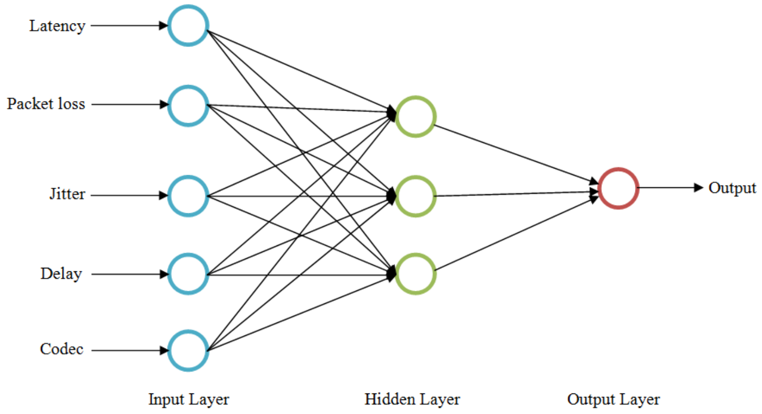


Figure 03: Feed forward architecture of RNN

Some degraded samples from the database are randomly chosen to train the ANN. Using ANN simulator [14] in Mat lab, ANN is trained. The remaining samples are used to test and the respective MOS values are recorded. The respective graph is plotted with reference to the PESQ MOS values and the ANN MOS values to show it visually.

IV. RESULTS OF EXPERIMENT

We have carried out an experiment using different codec for quality of service when data packets are lost. The figure 4 shows the graph between PESQ versus RNN MOS for codec alaw. The graph shows that steadily it degrades the quality as the percentage of packet loss increases. The figure 5 is for the codec GSM.

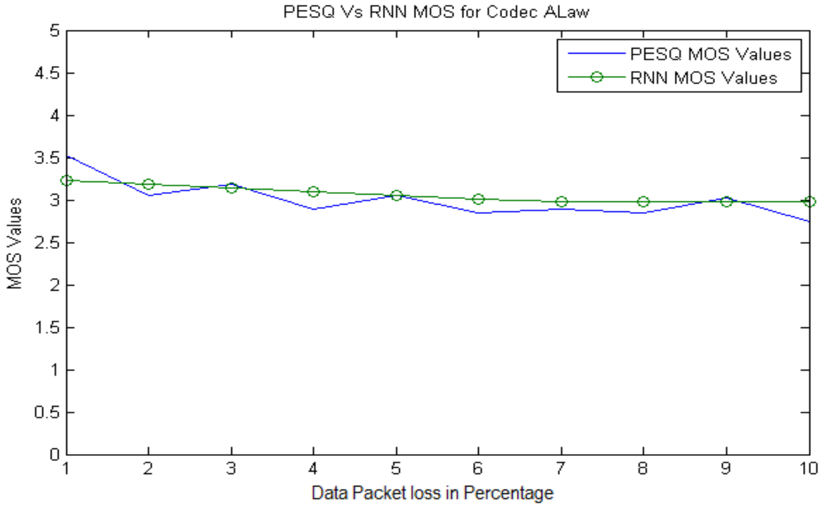


Figure 04: PESQ Vs RNN MOS using codec alaw

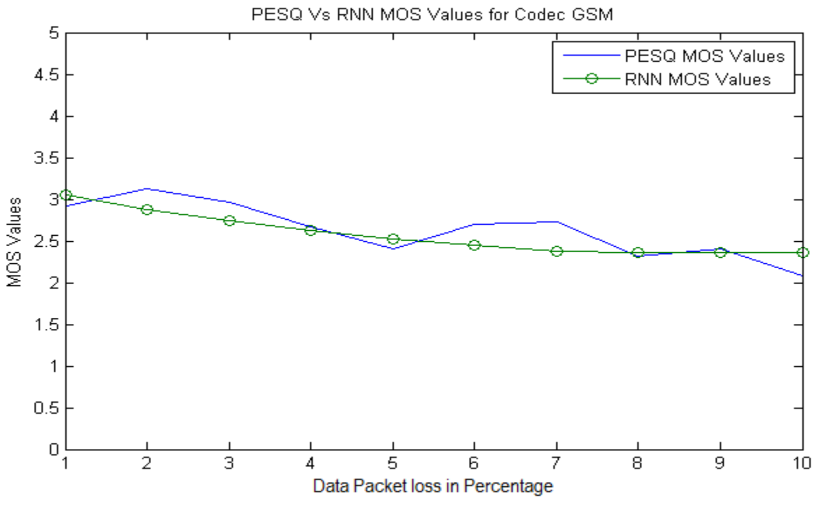


Figure 05: PESQ Vs RNN MOS using codec GSM

Here, it is seen that GSM is more sensitive to packet loss as compared to alaw. In figure 6 it is seen that for codec Speex the response is similar to alaw. It is also seen that the codec ulaw (Figure 7) gives good quality or small amount of packet loss but the quality gradually decreases with increase of packet loss. So, depending upon the situation and characteristic of network different codecs can be used efficiently for packet loss.

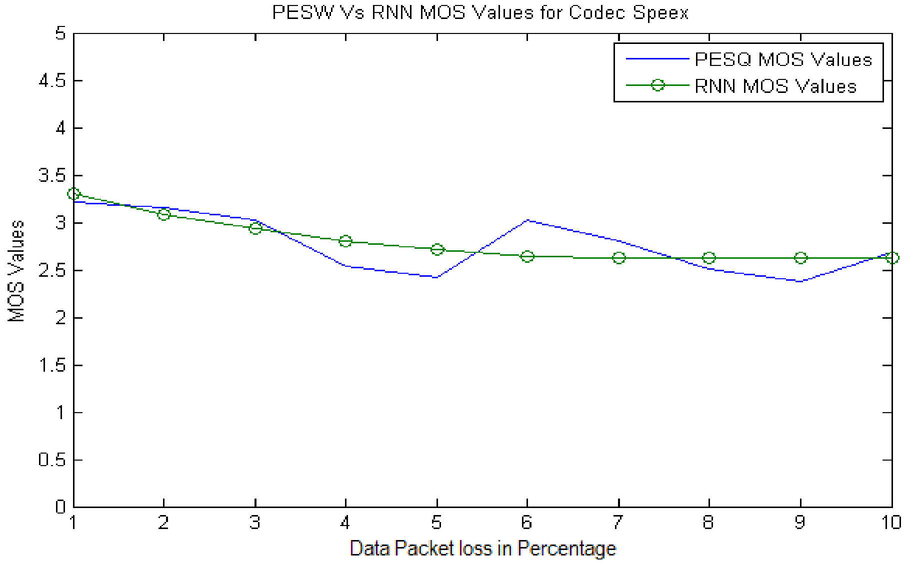


Figure 06: PESQ Vs RNN MOS using codec Speex

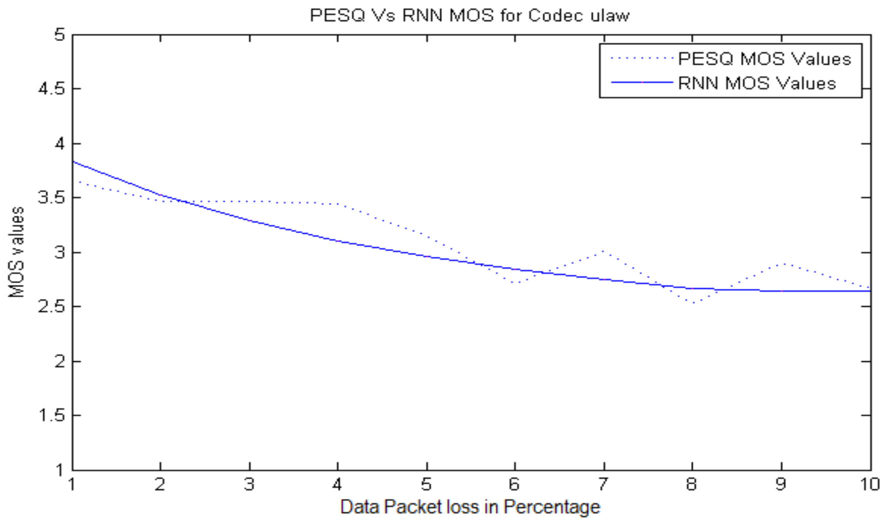


Figure 07: PESQ Vs RNN MOS using codec ulaw

When ANN trained properly it is observed that the plot of the ANN closely follows that o PESQ. We have calculated the Mean Square Error (MSE) of ANN to be 0.000458. This shows that ANN is predicting MOS quite accurately.

V. CONCLUSION AND FUTURE WORK:

In this paper, we presented a model to calculate MOS using ANN. The main purpose is to calculate the MOS without any reference of source / original voice signal. It can be calculated through network parameters. We calculated for different percentage of packet loss, keeping other parameters such as delay, jitter of network as constant. We have also calculated the MOS for different codecs.

In future, other parameters can also be included to improve accuracy.. Here, we used only Session Initiation Protocol (SIP). It can also be extended for different VoIP protocols. ANN can also be implemented along with the network at receiving end to get the QoS instantly.

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