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Achieving maximum VoIP Call Using Fuzzy Logic

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Abstract From the most beginning two main fundamental technologies are necessary for the evaluation & existence of VoIP i.e. telephone most widely used as first & second is the internet. In 1870sAlexander Gram Bell and Elisha Gray invented the telephone. The Internet is first developed by ARPANET (Advanced Research Projects Agency Network) in 1967, founded by the U.S. Department of Defense in 1957. As ARPANET provide a decentralized communications network would not be interrupted by a potential global war.

VoIP stand for Voice over Internet Protocol is a part of internet technologies. It is a communication Protocol that has transmission technologies for delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other technology that is synonym of VoIP i.e. IP telephony, voice over broadband (VoBB), Internet telephony, broadband phone and broadband telephony. VoIP has been progressively more popular in recent times because of its affordability, how-ever; poor reliability and voice quality remain significant factors that limit the extensive adoption of VoIP systems. Good voice quality is a vital factor for users transiting from the Public Switched Telephone Network (PSTN) to VoIP networks. It has been seen that line delay is one of the vital factors that depreciate voice quality in VoIP systems. Numerous non-real-time algorithms have been developed to estimate different aspects of voice quality in VoIP systems. In fact there is no real-time algorithm that estimates the delay content of a VoIP discussion, which could enable to take some curative actions to improve the quality while the call is in progress. In this paper, I propose a real-time fuzzy algorithm to estimate the VoIP call Quality in VoIP networks. The results obtained that shows the algorithm is able to track and estimate VoIP System performance in real-time. This algorithm could be embedding in VoIP systems to allow operators monitors calls in real-time.

1. Introduction

VoIP stand for Voice over Internet Protocol is a part of internet technologies. Internet telephony is used in Following Applications that are SMS, voice, voice-messaging and fax that are used Via internet rather PSTN (public switched telephone network).So the steps involved for originating the VoIP telephone call are signaling, media channel setup and digitization of the analog signal of Voice encoding, packetization and transmission is done by internet protocol (IP) packets over a packet-switched network. At the Receiver Side same steps are repeated but in reverse order i.e. reception and decoding of IP packets analog to digital conversion for regenerating the voice stream

In order to make sure good voice quality, several non-real-time algorithms that estimate the voice quality such as the Perceptual Evaluation of Speech Quality (PESQ), Per-ceptual Analysis Measurement System (PAMS), European Telecommunications Standards Institute (ETSI) and the Perceptual Speech Quality Monitor (PSQM) Computation Model (E-model) have been developed.

A major obstruction in carrying voice traffic over data networks is the increased delays encountered in these networks. Longer delays due to the busty nature of the IP net-work, call connectivity and speech distortion.

1.1. Theoretical Background

In most cases our daily conversations take place in the presence of delays. As delays are introduced by presence of echoes we hear echoes of our speech waves as they are reflected, from the walls and the floor, though, if the reflected waves reach your destination shortly after we spoke them, we do not distinguish them as echo but as some noise. On the other hand if the reflected wave takes 30 or 40 milliseconds (ms) to come back to us, we will identify it as an irritating echo. Echo is very closely related to other factors such as jitter, packet loss and delay that affect the voice quality of a VoIP call.

Delay is also depending upon the telecommunications network primarily by transmission equipment and transmission facilities. The delay could be significant or negligible. Depending on the type of trans-mission equipment used in the network and network topology, 30 ms of roundtrip delay can also occur in connections that are across country or just across town

Jitter is another issue encountered by voice calls; it is caused due to retransmission of lost packets, which is straight linked to delay in the IP network. Delay could also be caused by diverse factors that are already mentioned above.

Packet Loss can also occur for a variety of reasons; these include; traffic congestions, misrouted traffic and a link failure. In IP environment, the packets could be re-routed or retransmitted and this causes dela

1.2. Related Works

Many Researchers have also worked on evaluating the voice quality of a VoIP calls by improving on earlier methods mainly PESQ, the E-Model and PAMS. Modified E-Model [2][8], It is an introduction of a Packet-based Echo Canceller and others have been proposed, but none of them has been real-time. The major reason for the delayed success of the VoIP might be credited to the fact that the Internet was considered to be a fault tolerant data transmission /exchange medium, and the traffic re-routing was the main target in the case of Internet server additions and removals. The transmission delay or delivery time was never the primary concern in the design phase although the Internet Protocol (IP) has hold up for real-time transmissions [2]. As according to Periakarruppan et al.; [3], the general problem that was occurs when a VoIP network is utilized is jitter, delay, echo, and loss of packets. All this problems are very closely inter-linked.

As it was described by Ditech Networks [13], that there are three classes of object voice quality evaluation metrics: (i) psycho-acoustic metrics, (ii) elementary metrics, and (iii) network-parameter based metrics

(i) Psycho-acoustic metrics change voice signals to a compact representation to retain only perceptually significant aspects. These metrics also aim to predict the one-sided quality over a wide range of voice signal distortions. An example of such metric is the PESQ algorithm

(ii) Elementary objective voice quality metrics depend on low-density signal processing parameters and techniques to predict subjective voice quality. Elementary metrics usually have smaller correlations with subjective voice quality than highly complex psycho-acoustic metrics and also do not provide the opinion modeling needed for psycho-acoustic coder algorithm development. on the other hand, elementary metrics represent a good engineering tradeoff for networking and communication system and researchers and developers in that they can allow for fairly detailed conclusions about voice quality while having very low computational complexity

(iii) Parameter-based metrics do not critic the actual voice signal. In fact, these metrics sum impairment factors that explain the character components of the communication system. In the E-model the packet loss and delay in a VoIP system are transformed into impairment factors. Parameter-based metrics in this the E-model also hold promise for predicting such subjective voice quality but still also requires extensive refinements and verifications.

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Now we summarize the different aspects of voice quality evaluation in Figure 1.

$$\textit{Voice Quality}\begin{cases}\textit{Network Quality}\\ \textit{Subjective Voice Quality}\\ \textit{Objective Voice Quality} \rightarrow \begin{cases}\textit{Parameter - based metrics (PAMS)}\\ \textit{Psycho - acoustic metrics (PSQM)}\\ \textit{Elementary metrics (Proposed Method)}\end{cases}\end{cases}$$

Figure 1. Classification of voice quality algorithms for VoIP systems

To reduce the effect of delay in voice traffic and improve on user apparent QoS, it is necessary to approximate the delay content of voice traffic in real-time. Which could also help an operator or the system take a remedial action such as switching a user to a less delay channel while a call is in progress? We propose a simple objective low computationally complex algorithm using fuzzy inference system to estimate the delay content of the voice traffic over VoIP network in real-time. This proposed algorithm is to deal with the non real-time nature of algorithms mentioned at the beginning of this section.

2. Proposed Algorithm

In this section we will build up an algorithmic tool for evaluating the delay content of voice traffic using a fuzzy inference system.

The proposed fuzzy inference system algorithm estimates the delay section of the voice quality factors and also serves as a building block for a passive, objective, voice quality algorithm based on elementary metrics. However As we have discussed earlier, the vital issues in delivering good voice quality over IP networks are: delay, jitter, packet loss, and echo. These all issues are linked, but there is a stronger connection between packet loss, delay and jitter. As Jitter in VoIP systems is usually compensated for by using a payout buffer at the receiving end, which also introduces delay and packet loss.

There are different parameters which estimate delay. Therefore, we have taken parameters from real time system so the input parameters that we get from a standard are:

1. DNS Resolution Time(DNS-RT) which is delay between that apply for IP address for a domain Internet address and a valid reply from a Domain Name Server.
2. Roundtrip Delay (RTD) which is the Sum of the absolute delay on an outgoing and return path.
3. Speech Distortion (SD) which also indicate Normalized metric that is the indication of magnitude of unnatural sound not originally spoken by the caller.
4. Call loss Ratio (CLR) is the proportion of calls that fail to establish a connection due to network faults/congestion.

Based on these parameter I proposed the real time algorithm to estimate the VoIP call quality in VoIP network by using the parameter of call connectivity, Internet data & speech distortion As DNS Resolution Time (DNS-RT) & Roundtrip Delay (RTD depend on call connectivity Internet data, Speech Distortion (SD) depend on call quality distortion all these parameter are taken from matrix of Minacom reference Chart

We have described all the parameter mentioned above in figure 2 showing relation between DNS Resolution Time (DNS-RT), Roundtrip Delay (RTD), Speech Distortion (SD), and CLR. In this figure DNS Resolution Time(DNS-RT), Roundtrip Delay (RTD), Speech Distortion (SD) are act input parameter of delay of VoIP call quality in VoIP network andCall loss Ratio (CLR) act as output parameter of VoIP call quality in VoIP network.

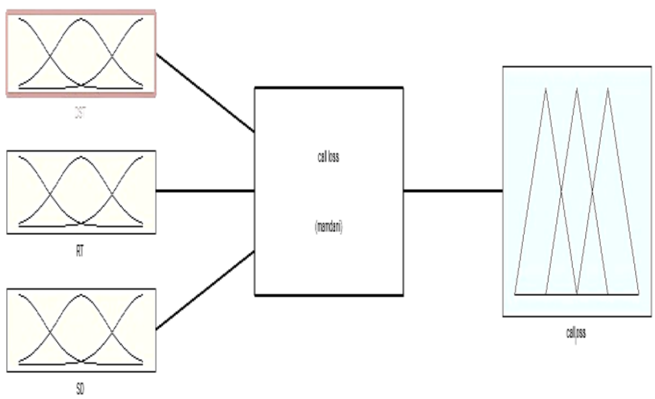

Figure2:-Fuzzy interference system to estimate VoIP call quality in VoIP network.

To achieve a real-time low computationally complex algorithm, we have to drive output membership functions of each parameter using fuzzy interference system showing the effect on the output.

Table no. 1 shows the fuzzy sets associated with the input parameters and effect on proposed.

So we have defined three output membership functions of which will give an estimate of the delay content of voice traffic can be described by the following equations:

The membership function for the “Good DNS-RT” fuzzy set.

$$U_{GDNSRT} = \begin{cases} \frac{100-x}{100} & 0 = x = 100 \\ 0 & x > 100 \end{cases}$$

The membership function for the “Moderate DNS-RT” fuzzy set.

$$U_{MDNSRT} = \begin{cases} \frac{x-100}{200} & 100 < X < 300 \\ \frac{400-X}{100} & 300 < X < 400 \end{cases}$$

The membership function for the “Bad DNS-RT” fuzzy set.

$$U_{BDNSRT} = \begin{cases} \frac{X-400}{200} & 400 < X < 600 \\ 1 & X > 600 \end{cases}$$

Table1. Fuzzy sets associated to the input parameters

Fuzzy Set	Description
Good DNS-RT	Represents values of DNS Resolution Time that the output is good as delay is less
Bad DNS-RT	Represents values of DNS Resolution Time that the output is good as delay is more
Moderate DNS-RT	Represents values of DNS Resolution Time that the output is good as delay is moderate

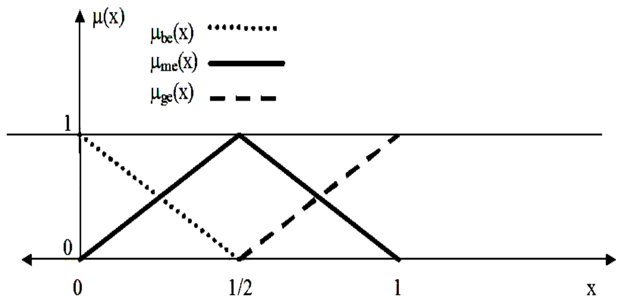

Figure3. Graphical representation of the output membership functions for delay content

The approach that we have used is to define the input membership functions based on standards that we have got from Minicom matrix chart, and then we spent some time for modification of those membership functions. But the modification is done after the fuzzy inference system is designed as we require using the output of the algorithm as a feedback for tuning. In the next paragraph we describe the proposed fuzzy inference system with the complete set of fuzzy rules, defuzzification and operations.

The fuzzy rules collectively with the fuzzy membership functions are the main fundamentals that used triangular method for the proposed fuzzy inference system. We have describes the some fuzzy rules that we have used for our proposed algorithm to define the following membership functions for each one of the fuzzy inference input and output variables.

Fuzzy Rule:-

1. If DNS Resolution Time (DNS-RT) is bad then CLR is bad.
2. If Roundtrip Delay (RTD is bad then CLR is bad.
3. If RTD is good SD is moderate then CLR is moderate.
4. If DNS-RT is good and RTD is good and SD is Good then CLR is good
5. If DNS-RT is bad or RTD is bad or SD is bad then CLR is bad

We have used MATLAB and its fuzzy logic toolbox to implement our proposed algorithm and ran a set of 10 calls. We have implemented it in a way that we could feed MATLAB 2008a version with diverse fuzzy inference systems at a time. The variation among the fuzzy systems was only in terms of the membership functions. All systems had the same input fuzzy rules, fuzzy operations, and variables, but different membership functions for each input and output fuzzy variables. As we have taken 10 calls based on these we will evaluate voice traffic.

As there are so many parameters that can be used to define delay content .And we have define three membership function using specific triangular membership that has a positive and a negative slope As a result we have derived the following membership functions for the fuzzy sets.

The membership functions for the "Good RTD" fuzzy set.

$$U_{GRTD} = \begin{cases} \frac{100-X}{100} & 0 \le X \le 100 \\ 0 & X > 0 \end{cases}$$

The membership functions for the "Moderate RTD" fuzzy set.

$$U_{MRTD} = \begin{cases} \frac{X-100}{150} & 100 \le X \le 250 \\ \frac{400-X}{150} & 250 \le X \le 400 \end{cases}$$

The membership function for the "Bad RTD" fuzzy set

$$U_{BRTD} = \begin{cases} \frac{X-400}{200} & 400 < X < 600 \\ \frac{400-X}{150} & X \ge 600 \end{cases}$$

The membership function for the "Good SD" fuzzy set

$$U_{GSD} = \begin{cases} \frac{4.6-X}{4.6} & 0 < X < 4.6 \\ 0 & X > 4.6 \end{cases}$$

The membership function for the "Moderate SD" fuzzy set

$$U_{MSD} = \begin{cases} \frac{X-4.6}{2} & 4.6 < X < 6.6 \\ \frac{7.5-X}{0.9} & 6.6 < X < 7.5 \end{cases}$$

The membership function for the "Bad SD" fuzzy set

$$U_{BSD} = \begin{cases} \dfrac{X-7.5}{2} & 7.5 \le X \le 9.2 \\ 1 & 9.2 \le X \le 10 \end{cases}$$

The above mentioned membership function are input parameters and now we will define membership function for output parameter.

The membership function for the “Good CLR” fuzzy set

$$U_{GCLR} = \begin{cases} \dfrac{10-X}{10} & 0 \le X \le 10 \\ 0 & X>10 \end{cases}$$

The membership function for the “Moderate CLR” fuzzy set

$$U_{MCLR} = \begin{cases} \dfrac{X-10}{3} & 10 \le X \le 13 \\ \dfrac{15-X}{2} & 13 \le X \le 15 \end{cases}$$

The membership function for the “Bad CLR" fuzzy set

SD, RTD and the output is evaluated from these para

Table2. Average reading of 10 VoIP Calls

CALL No	DNS-RT (MS)	RTD (MS)	SD (%)	Output	Quality
1	450	800	4.3	58.988	Bad
2	450	200	4.3	58.099	Bad
3	50	100	4.3	50	Bad
4	50	90	4.3	4.5	Good
5	90	90	5.2	12.5	Moderate
6	500	50	7.2	56.987	Bad
7	50	50	4.2	4.500	Good
8	470	110	6.2	58.212	Bad
9	40	60	2.2	3.823	Good
10	110	50	7.2	12.5	Moderate

As shown from the table we have recorded the 10 VoIP call and the parameter that are used to show the delay in VoIP call quality in VoIP network the output shown from these parameter that is call loss ratio changing according to the different value of the DNS-RT, RTD and SD. According to the different value of the above parameter we can evaluate the VoIP call quality in VoIP network.

Figure 3:-MATLAB surface view for DST-RT and RTD

$$U_{BCLR} = \begin{cases} \dfrac{X-15}{85} & 15 \le X \le 20 \\ 1 & 20 \le X \le 100 \end{cases}$$

3. Performance Evaluation

Now to estimate the performance of our proposed real-time delay detection approach, numerous simulations were performed using MATLAB for a one–to–one VoIP voice call situation. So in this section, we will present a detailed explanation of the estimate process and results obtained.

In the proposed algorithm we have taken 10 calls with different levels of voice quality. All these calls were all earlier generated during a live VoIP call and recorded in PCM format. We then collected the measurements from this calls and analyzed these measurements and finally used them as inputs to our fuzzy algorithm. As all these parameters are described in the previous section, and the collected measurements from the recorded call that were the DNS-RT,

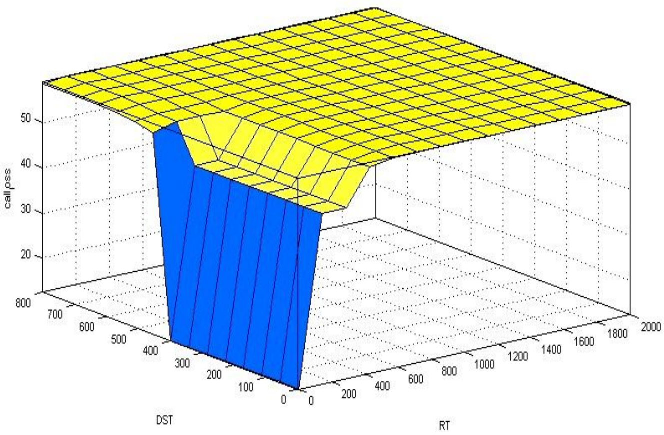

Figure 4:-MATLAB surface view for SD and RTD

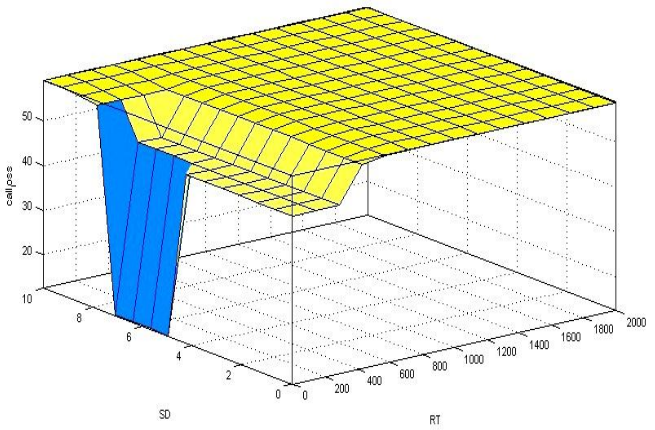

Figure 3 shows a 3-D view of the delay content, showing MATLAB surface of DNS-RT and RTD and their relationship with Call loss. We can conclude from these results that the proposed algorithm correctly tracks and estimates the delay content also defining the range and quality of voice traffic in a VoIP system.

Figure 4 shows a 3-D view of the delay content, showing MATLAB surface of SD and RTD and their relationship with Call loss. We can conclude from these results that the proposed algorithm correctly tracks and estimates the delay content also defining the range and quality of of voice traffic in a VoIP system.

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5. Conclusions

This effort presented an algorithm which is a basic element of an objective, voice quality passive, algorithm that can run in real-time and estimates the voice quality for live calls in VoIP systems. Also the use of fuzzy logic was provoked by its ability to give reasonably good result with low computational complexity. This proposed algorithm can run and give consequences in real-time in the surrounded system that processes the VoIP calls, then by giving operators an almost immediate estimation of the quality of their network with respect to delay without the need for a reference signal.

REFERENCES

[1] In 2007 S. Tartarelli, J. Quittek, S. Niccolini, M. Stiemerling, M. Brunner, T. Ewald from NEC Europe Ltd., Kurfürsten-Anlage 36, 6915 Heidelberg, Germany, described about the email spam problem, which could evolve in the next years towards a problem affecting the next contemporary's telephone system based on IP

[2] Ren, Jiuchun;, WeiChao ,ChongMing; Mao, Dilin, Zhang (2010) Enhancement to E-Model on standard deviation of packet delay,2010 3rd International Conference, vol. 23,24,25, pp.256-25

IP convergence network.

[3] Periakarruppan, G.; Azhar, H.; A.L.Y.; Rashid, A., (2006) Packet Based Echo Cancellation for Voice Over Internet Protocol Simulated with Variable Amount of Network Delay Time,. 2006 IEEE Region 10 Conference, pp 1 to 4.

[4] Adaptive Digital Technologies, (2010) The Echo "Pheno-menon", Solutions and Causes, http://www.adaptivedigital.com/product/echo_cancel/echo_explain.htm.

[5] In 2006 Rainer Baumann, Stephane Cavin and Stefan Schmid in University of Berne, recommended that VoIP world is evolving rapidly and its security issues are well discussed among researchers

[6] In 2008 Ying-You, Wen and Zhao Hong, The Software Center of Northeastern University, Shen Yang, China anticipated to resolve the SPIT Spam over Internet Telephony security risk in the ALL-

[7] Thompson, R. and Ngamwongwattana, B. (2010) Sync & Sense: VoIP Measurement Methodology for assess One-Way Delay without Clock Synchronization- Instrumentation and Measurement, vol. 59, pp 1319 – 26

Applications (ICCEA), 2010 Second International Conference, vol.1, pp 612 – 15.

[9] Chien-Chung Shen and Justin Yackoski , (2010) Managing End-to-End Delay for VoIP Calls in Multi-Hop Wireless N/w s, IEEE Comm. Society subject matter experts for publication in the IEEE INFOCOM 2010 proceedings.

[10] Lingfen Sun and Emmanuel C. Ifeachor (2006) Perceived Speech Quality Prediction fsor Voice over IP-based Networks, Department of Electronics' Communication and Electronic Engg., University of Plymouth, Plymouth PL4

[11] Petri, D. and Paglierani, P. (2009) Uncertainty valuation of Objective Speech Quality Measurement in VoIP Systems, Measurement and Instrumentation , vol.58, no.1, pp.46-47

[12] Ross, T. (2004) Fuzzy Logic sets with Engineering Application willey (2004

.

[8] Dr.Sarabjeet Singh ,Harjit Pal Singh Dr.Jasvir Singh, (2010) Computer Modeling & Performance study of VoIP under Different planned Conditions, Computer Engg. and

[13] Ditech Networks. Echo Basics Tutorial including echo cancellers and echo's effect on QoS" (2010) http://www.dite chnetworks.org/learning Center/echoBasics.html.

[14] Wai-Yip Chan and Falk, T.H. and. (2008) Hybrid Sig-nal-and-Link-Parametric Speech Quality Measurement for VoIP Communications, Speech, Audio, and Language Processing, vol.16, no.8, pp.1579-89

meter that is CLR. As Shown in table No