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Dynamic Bandwidth Estimation and Utilization for Voice Traffic in Wireless Networks

M.Prashanthi
Assistant Professor (Adhoc)
JNTUA College of Engineering Anantapur

***Abstract*: Nowadays the demand for voice calls and VOIP is rapidly increasing. Suppose in saturated conditions around 2.5 Mbps of voice data can be generated with a 256-kbps data rate and it is difficult to handle this amount of burst transmission. If the network is saturated means around 2.5 Mbps data is generated it takes more time to transmit all the packets. In this paper, a dynamic bandwidth estimation and utilization is proposed for Voice data that enables voice to transmit more packets within stipulated time.**

Keywords: Voice, bandwidth, saturated conditions, traffic

I. INTRODUCTION

Voice quality is a critical issue to be satisfied with voice-based applications. VoIP is an emerging protocol over PSTN which is used in general telephone system. Transmission of voice over the internet associated with several issues like QoS, security, etc. One among all issues is bandwidth. Some poor-quality voice calls may be terminated in the middle of the communication. General user can't understand why it has happened and how can do avoid it.

While many things are wrong with a phone call there is not enough bandwidth to holdup the channel, which is a very common problem. If the connection is not good, then a break may occur in calls and the call quality also becomes worse. VoIP is the latest technology, which is able to perform several tasks from phone calls to video conferencing. For now, its development wings are not spread much wider because of its sensitivity.

Demand for Voice quality in internet protocol is rapidly increasing and cannot be separated from PSTN quality. The voice quality of VoIP is much competing with PSTN quality nowadays. The main issue is bandwidth because it is a primary resource for everything on the internet. This should be enough for voice protocols. It should have some space for each phone call. The VoIP system does not meet the demands and the quality of the call is low, as the system becomes crowded with many cells in a limited bandwidth. Voice data will be processed on the internet in terms of packets. Raw voice data converted into digital packets by codec. Many codec models are available in the market and this codec decides how much Bandwidth is required for voice call. Different codecs require different bandwidths like 48,64,128 etc.

Similarly, devices on your network, for instance, the available bandwidth can be limits by the routers. It should be specified appropriate required traffic in the network. Usually assume that voice and data are given equal priority in our network, which is the wrong assumption. Due to its sensitivity, voice flow should be given high priority when all flows are in the traffic.

Most widely used codecs are:

Codec type	Minimum Bandwidth required	Description
G.711	64	This is one of the oldest codecs around (1972). Very low processor requirements. In case of high bandwidth, it works best, which makes it some little bit antiquated to the Internet but still good for LANs. It needs a speed of a minimum of 128 kbps for two-way.
G.722	48/56/64	Produces results with better quality and clarity by capturing frequency ranges that are double and as large as G.711, It is very close to or even better than that of PSTN. Network congestion conserves the bandwidth. It adapts to variating the compressions.
G.723.1	5.3/6.3	It runs in low bandwidth environments and with dial-up Because it has been worked under a low bit rate. However, it requires more processor power, more compression with high-audio quality.

G.726	16/24/32/40 kbps	This works under 16,24,32,40 kbps.
G.729	8	This one is improved than others which are G-family and also it is licensed. It utilizes tremendous bandwidth. Error tolerant. End users pay indirectly while purchasing handsets or hardware that enforces this license.
GSM	13	It is available on many software and hardware platforms because it is free. It has a high compression ratio. It provides a MOS of 3.7, which is good. This is being used in GSM telephones.
iLBC	15	This is referred to as Internet Low Bit Rate Codec. Now, it is free because it has gotten by Google. Mostly used by many VoIP apps those are with open source. It is Robust to packet loss.
SILK	6 to 40	This code was introduced by Skype. It is also open-source silk is true base codec for another code opus. Whatsapp using the Opus codec.
Speex	2.15 / 44	This is the most widely used codec because it minimizes bandwidth usage.

II. PROPOSED METHOD

Consider two end points of a channel of a generic network, one of them acting as a *transmitter* while the other one acting as a *receiver*. We may think of a channel as a link of a network or a path composed of more than one link. The transmitter generates and sends a short packet to the receiver once every *T* seconds. The receiver, upon reception of a packet, computes its inter arrival time, which will be used for an estimate of the load of the channel. In such a framework, if the channel is lightly loaded, it is reasonable to expect that also the receiver will receive asynchronous (or quasi-synchronous) train of packets. On the other hand, as the load of the channel increases, we expect to see, at the receiver site, a degradation of such a synchronicity.

These measure packets do not need to carry information and, therefore, their size can be very small. Indeed, if *T* is chosen properly, the train of packets does not significantly alter the overall load of the channel. This way of estimating the load has several advantages. First, there is no need of polling the nodes of the channel (e.g., routers) in order to get load information: not everyone, in fact, can normally access routers and, when access rights are given, routers might have different access protocols (e.g., SNMP, CMIP). Second, this way of estimating does not require to know almost anything about the channel. In particular, we do not need to know the number and type of active connections on the channel in order to come up with an estimation. This estimating mechanism provides a way of measuring directly the part of the channel still available, the free part of it.

Each valuable real time service (e.g., video, voice) has the need of a certain degree of synchronicity as a main Quality of Service requirement. By mapping the degree of synchronicity of the train of packets at the receiver for each service, we are able to say whether such a service can fit into the channel or network.

It is worth noting that not every source-destination pair need generating the test train of packets connections, even though the length of a test packet is of the order of few bytes. One possible alternative is to have a pair of "multimedia servers" performing this operation and distributing the results of the analysis, either periodically (with period much larger than *T)* or on demand. In this case, the additional load caused by the measurement packets could really be negligible. The presence of such "network elements" performing the estimation would also have the advantage of by-passing possible source control mechanisms that might alter the statistics.

III. SIMULATIONS

Simulations are conducted using ns2.35. Fifty nodes are generated and placed in random positions. All nodes are roaming using the random waypoint mobility model. 10 nodes behave as source nodes, and 10 are destinations. All traffic like voice, video and best effort are generated by each source node. Approximately every node can generate around 1 Mbps voice and video data as per the above rates. According to that, every station generates 128 bytes of BE and BK traffics. So that the network is saturated with all types of traffics.

Based on the above criteria, 10 nodes can generate around 11 Mbps data. The delay parameter is considered to estimate the performance the proposed the system.

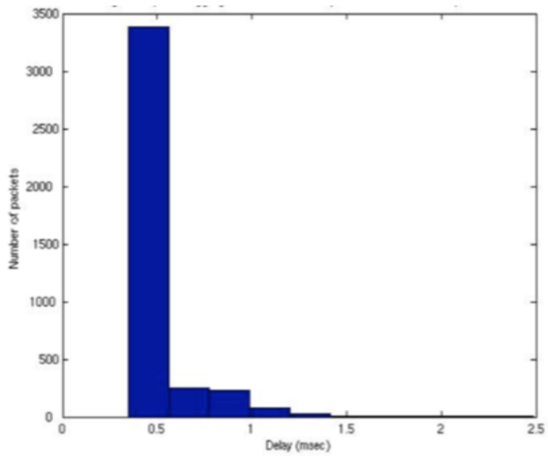

Figure 1: Number of packets delivered at various delay values

The above figure shows that almost all voice packets have been delivered at 0.5 msec delay which is almost negligible.

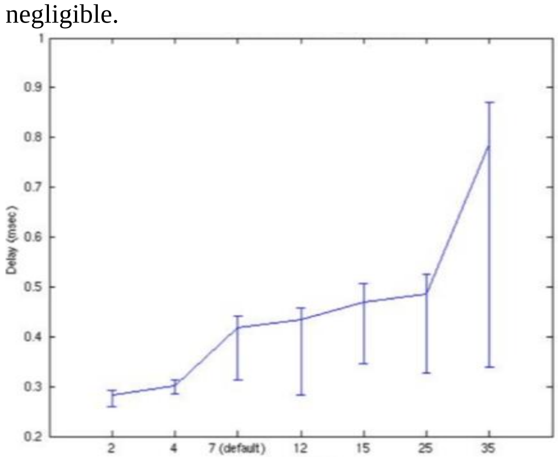

Figure 2: Mean packet delay for voice flows

The mean packet delay for any voice packet is very less i.e almost 0.9 msec which is also negligible for any real time service

The minimum packet delay for all the voice packets is around 0.4msecs. However the figure 1 shows that all the voice packets have been delivered with very short delay.

IV. CONCLUSION

A mechanism for estimating the available bandwidth between two end points of a network has been presented. The purpose of this mechanism is to carry out the estimation procedure with no information the overall offered load. Focus has been primarily put on real-time services even though also non-real-time ones often have maximum delay requirements. The mechanism works by performing, at one side of the

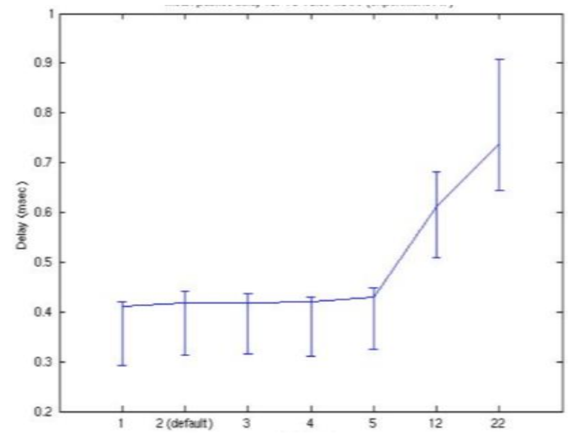

Figure 3: Minimum packet delay for voice flow network, a set of packets synchronously transmitted by the other end point.

REFERENCES

1.Yi-Ting Mai, Chih-Chung Hu,-Design of dynamic resource allocation scheme for real-time traffic in the LTE network, Journal of Wireless Communications and Networking, March 2022

2. Hyun-Woo Kim, Jun-Hui Lee, Yong-Hoon Choi,-Dynamic bandwidth provisioning using ARIMA-based traffic forecasting for mobile WiMAX, Computer Communications, Vol. 34, Issue 1, pp 99-106, January 2011.

3. JC. Bolot,-Characterizing End-to-end Packet Delay and Loss in the Internet. *Journal of High-Speed Networks*, vol. 2, no. 3, pages 305-323, July 1993.

4. N. Seiz, S. Wolf, S. Voran, and R. Bloomfield,-User-oriented Measures of Telecommunication Quality. *IEEE Communications Magazine*, pages 56-66, January 1994.

5. D. Ferrari,-Client Requirements for Real-Time Communication Services. *IEEE Communications Magazine*, pages 65-72, 1990.

6. W.E. Leland, M.S. Taqqu, W. Willinger, and D.V. Wilson. On Self Similar Nature of Ethernet Traffic (Extended Version). In *IEEE 1hmsactions on Networking*, pages 1-15, Oct 1993.

7. A. Erramilli. Chaos, Fractals and Traffic. In *Proceedings ofthe 6th MultiG Workshop*, Electrum, Stocki: JlmKista.

8. A. Erramilli e W. Willinger. Fractal Properties in Packet Traffic Measurements. In *Proceedings of Seminar*, St. Petersburg.

9. A. Erramilli, 0. Narayan, and W. Willinger. Experimental queueing analysis with long-range

dependent packet traffic. *IEEE Transactions on Networking*, pages 209-222.
10. B. Mandelbrot. *The Fractal Geometry of Nature.* Freeman, New York.