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Simulation and Analysis of Impact of Buffering of Voice Calls in Integrated Voice and Data Communication System

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Abstract - Queuing theory and Markov chain analysis plays vital role in analyzing real-life problems. It is applied to wired network, wireless network and mobile communication to analyze the packet traffic in packet switched network. In this simulation and analysis, integrated communication system such as voice and data is simulated with different queue size for voice calls with different arrival and service rate and its results are analyzed to study the impact of buffering of voice and data calls for the proposed integrated wired network using Queuing theory and Markov chain analysis. We also propose to optimize the system characteristics in an attempt to provide better Quality of Service (QoS) for systems with integrated voice and data calls. The proposed models have two traffic flow namely voice calls (real-time traffic like audio) and data calls (data traffic like FTP). A single channel is assigned for voice and data calls. The incoming voice and data calls are queued when the channel is busy. Voice calls are delay-sensitive therefore priority is assigned to Constant Bit Rate (CBR) traffic voice request. For such systems, it is important to analyze the impact of buffering the voice calls as well as data calls for various mean arrival rates and mean service times for voice and data call requests. The impact of buffering the voice calls with different queue size, mean arrival rates and service rate are analyzed. These results of dedicated integrated voice and data communication system can be used for simulating any type of wired network. The minimum buffer or jitter required for both the traffic is calculated using Packet Delivery Fraction (PDF).

Keywords - arrival rate, packet delivery fraction, performance, queue size, QoS, voice and data traffic.

I. INTRODUCTION

The robust growth of Internet technology, coupled with the relatively low deployment cost of Internet Protocol (IP) networks, has created a push for an integrated "IP-based core" - a single network for data, video and voice access. However, the diverse service-requirements and novel traffic characteristics of the emerging Internet applications have posed many technical challenges that the Internet community must address in the near future, as the emerging multimedia applications begin to constitute an ever-increasing fraction of Internet traffic. High quality interactive voice and video applications tolerate little delay variations and packet losses. The economic advantages of provisioning this range of service by means of a single infrastructure are considerable. Effective networking for this diverse range of multimedia applications requires in-depth research in various fields of the internet based application. The change to VoIP provides a more cost effective communication solution to the business and more and more businesses are developing IP based

solutions for their day to day use. This can be extended to the dedicated applications that involve both voice and data traffic.

Voice calls are delay sensitive. Therefore, the choice of buffer management is very much essential to enhance the overall performance of integrated communication system. The significance and the impact of buffering on the QoS parameters for each traffic type is simulated and analyzed in depth in an integrated communication channel. Two different types of traffic voice and data calls with random inter-arrival and service time are analyzed. For finite queuing capacity arrival rate to the queuing system depends upon the number of packets being served and waiting in the queue [6]. Impact of buffering of voice calls in integrated voice and data services where arrivals using Batch Marked Markov chain process and service times distribution is phase type has improved performance of integrated system [1]. The proposed network is simulated for different arrival rate and service time with finite buffering of voice and data calls.

II. DESCRIPTION OF PROBLEM

In Fig.1, the proposed system already has radio link that exist between Master and six Slave units that sends voice and data. The wired link sends and receives only data on communication channel that exists between Master Post and six slave stations. Master post and Slave units are connected to Modem .Currently there are six Slave units and one Master Post. Modem has interfaces RJ-45 and Serial communication for data communication and Radio link for voice and data. Master post and Slave units are connected in star topology. There is no communication between Slave units. All Slave units must have to connect to the Master post. The proposed integrated system must send messages (data) and voice over cable that exists between Master Post and six Slave units. The Time division Multiple Access (TDMA) technique is used in proposed system .It allows several users to share the same frequency channel by dividing the signal into different time slots. Users transmit in rapid succession one after the other each using its own time slots. In TDMA, Master or Master Post allocates contention free transmission to other slave or Slave units in the proposed system. Only one station can transmit at time thus avoiding collision. The system assumes the total bandwidth of network is 9600 bps.The size of data packet and voice packet has to be simulated and analyzed with finite queue size or buffer size with random arrival rates and service time for voice and data traffic in the integrated communication system to effectively design the proposed system. The performance of this network depends upon the design of jitter or buffer for voice traffic with different priorities for data and voice traffic. It assumes various priorities for voice and data traffic, Voice with highest priority, both equal and voice with lowest priority in order to meet the user requirement. The proposed system simulates finite as well as infinite queuing capacities. The proposed system is simulated using Network Simulator ns-2.

Fig. 1. Proposed voice and data communication system

III. SIMULATION SETUP

Fig. 2. NAM Animator

Ns-2 is a discrete event driven network simulator extensively used by network researchers to study the various parameters [7]. To investigate possible solutions for improving QoS, a number of simulations were undertaken using Transmission Control Protocol (TCP) flows as well as User Datagram Protocol (UDP) flows. TCP flows were used mostly for the shortest transmission and normal data flow whereas UDP was used for the real-time traffic flow. TCP by design backs off when TCP packets get dropped and retransmits the packets again.UDP by design transmits the packets once and never retransmits the packets again. Voice communication does not require retransmission as the delay would lower the quality of service and in the case of integrated data and voice system to ensure quality of service we need to ensure a smooth transmission of voice calls. Constant Bit Rate (CBR) traffic is generated for real-time audio traffic and File Transfer Protocol (FTP) traffic is generated for discrete data calls. Queue size is varied for different mean arrival rate and service rate for voice calls and data calls.

In Fig. 2, $n0$, $n1$, $n2$, $n3$, $n4$, $n5$, $n6$ are seven stations or nodes. In this experimentation setup, the node $n0$ is Master post and $n1$, $n2$, $n3$, $n4$, $n5$, $n6$ are six Slaves that are connected in a star topology. Station $n0$ (Master post) and station $n1$ (Slave 1) are connected using a bidirectional link that has 1ms of propagation delay and capacity or bandwidth of 9600 bits/sec. buffer or queue is assigned to bidirectional link between station $n0$ and station $n1$. Similarly stations $n0$ and $n2$, stations $n0$ and $n3$, stations $n0$ and $n4$, stations $n0$ and $n5$, stations $n0$ and $n6$ are configured like station $n0$ and $n1$. CBR (Constant Bit rate) traffic flow for real-time voice calls is attached to the source node $n0$ which uses UDP protocol to transfer the traffic to destination node $n1$.FTP traffic flow for data calls is attached to node $n0$ which uses TCP protocol to transfer data traffic to destination station or Slave1.here Slave1 is the TCPSink

and UDPSink which act as a receiver. Traffic Flow colors are given to both the traffic mainly blue color for CBR on UDP and red color for FTP traffic on TCP. Both the traffic flow are generated at the same stations. Let us assume the Master post ready to send data and voice traffic to one of the station Slave1. Since the Master post is in Master mode, it always acts as transmitting station and all other stations as receiving station and in TDMA, Only one station can transmit at time thus avoiding collision. DropTail queue or buffer management algorithm that implements FIFO is used. There are DropTail buffers at the head of every link in the network.

The focus in this experiment is on size of the buffer, inter- arrival time for voice calls and data calls with priority. Three different simulation experiments were executed using the ns-2 network simulator software. These three simulations were based on the same basic model. The tests were performed by varying the queue size for the different inter-arrival time for voice calls and data calls in an integrated service in the network. In Experiment 1, only CBR traffic is generated on UDP for a voice calls with different queue size, inter-arrival rate for CBR traffic and service rate for voice calls only. In experiment two, only FTP traffic is generated on TCP for a discrete data calls with different queue size, inter-arrival rate for FTP traffic and service rate for data calls only. In Experiment 3, integrated traffic, CBR is generated on UDP for a voice calls and FTP traffic is generated on TCP for a discrete data calls with different queue size, inter-arrival time for CBR and FTP traffic and service time for voice and data calls. The buffer size is fine tune to achieve improved performance. The size of Data Packet is 500 bytes and size of voice packet is 150 bytes. Simulation results are obtained by filtering the generated trace files using PERL scripts and AWK scripts.

IV. RESULTS

In Experiment 3, the voice traffic is generated to arrive at the interval of 1ms, 150ms, 1sec and data traffic is generated at 1ms ,service rate for both the traffic is 1ms for the jitter or buffer sizes (no. of packets) is varied from 0 to 200 for the proposed system.

Packet Drop

In Fig.3, the drop rate or packet loss for CBR packets on UDP is more than FTP traffic on TCP for the inter-arrival time of voice traffic is 1ms and the inter-arrival time of data traffic is 1ms. In fig 4, the inter-arrival time for CBR traffic has been increment to 150ms and FTP traffic 1ms ,This Fig. Shows that CBR

packet drop rate has decreased drastically and at queue size of 50 for both the CBR and FTP traffic and packet drop is almost zero .In fig 5. Further the inter-arrival time for CBR packets is decreased to 1sec with the 1ms inter-arrival time for FTP traffic, the packet loss is almost zero at buffer size of 30.

Fig.3. Queue Size vs. Packet loss

Fig. 4. Queue Size vs. Packet loss

Fig. 5. Queue size vs. Packet Loss

Packet Delivery Fraction (PDF)

As expected the Packet delivery fraction for CBR traffic is increasing with the increasing queue size or buffer length, however it has not reached up to the mark as expected with the increasing queue size, when CBR traffic is arrived at the interval of 1ms and the FTP traffic is arrived at the interval of 1ms and the service time is 1ms, as shown in Fig. 6.

Fig. 6. Queue Size vs. PDF

In Fig.7, Packet Delivery Fraction is high (100%) for both the traffic CBR and FTP at Queue size or jitter buffer 50. In this case, CBR traffic is arrived at the interval of 150ms and the FTP traffic is arrived at the interval of 1ms and the service time is 1ms.

Fig. 7. Queue Size vs. PDF

Packet Delivery Fraction is also high (100%) at queue size 30 for both the traffic as shown in Fig.8. In this case, inter-arrival time for CBR traffic is 1second and inter-arrival time for FTP traffic is 1ms and the service time is 1ms.

Fig. 8. Queue size vs. PDF

Average End to End Delay

In Fig.9, Average End to End delay increases with increasing queue size for CBR traffic but it remains constant for larger queue size for FTP traffic, When CBR traffic is generated at the interval of 1ms and the FTP traffic is arrived at the interval of 1ms and the service time is 1ms.

Fig. 9. Queue Size vs. Avg. E to E Delay

V. CONCLUSIONS

The simulation results show that voice call and data calls can be buffered with different inter-arrival time and service time for the both the traffic. Finite Buffering of voice calls improves the Packet Delivery Fraction for the given simulated network. This can be generalized for simulating networks with higher bandwidths for different mean arrival rate and service rate for the voice and data traffic with the increasing buffer length. This can be useful for designing voice and data packet size with the estimation for buffer size or jitter for any kind of VoIP wired network. These results can be useful for designing and developing any dedicated integrated defence applications and business VoIP networks.

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