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A Queuing Model for Congestion Control and Reliable Data Transfer in Cable Access Networks

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***Abstract*-A set of simple queuing theory models which can model the average response of a network of computers to a given traffic load has been implemented. The impact of variations in traffic patterns and intensities, channel capacities, and message protocols can be assessed using them because of the lack of fine detail in the network traffic rates, traffic patterns, and the hardware used to implement the networks. In order to access the networks it's a two stage sequential procedure for data transfer from a node at customer's premises to a central node. The stations which have data to transmit request will undergo contention stage and requests that are successfully received by the central node enter into the second stage. All the data will reside in a queue until they are scheduled by a centralized scheduler. Once the stages are set properly and all the data is queued accordingly then a congestion free communication will happen in the cable access networks. To establish a congestion free as well as reliable data transfer in cable access networks we are working with a access network which is hybridized with queuing models which has the capacity to maintain the data packets in a scheduled manner. We test on static as well as distributed (random) data which will arrive at the centralized node. From then its up to mathematical queuing model to route the data reliably without any collision free. The performance measures and various data transfer patterns are observed in this work.**

***Keywords:* cable access networks, queuing models, contention and congestion, routing strategies.**

I. INTRODUCTION

Emerging access networks like residential cable networks with burst speeds up to 40Mb/s - more than 2000 times faster than an ordinary dial-up modem - will most likely provide the next-generation data communication services, including internet access to homes and small business. These networks are currently being standardized (e.g. DOCSIS, IEEE, DAVIC/DVB) and therefore they are the focus of extensive research activity. Characteristic for access networks is a two-stage sequential procedure for data transfer from a station at the customer's premises to a central node. First, a contention stage is carried out (with other stations), in which stations that have data to transmit request a number of data slots. Requests that are successfully received by the central node enter the second stage: they queue in a request queue, until they are scheduled by a centralized scheduler. When scheduled, the station can transmit the data collision-free. Multimedia pertains to the interactive use of audio/video material, possibly enriched with text and graphics. video on demand is one of the most demanding services because of the huge storage and bandwidth requirements as well as real-time requirements in an interactive setting. The prospect of delivering VOD with instant access, interactivity, and browsing possibilities, comparable to that offered by a conventional video cassette recorder, to the homes of millions has attracted extensive interest from academia as well as industry. In the past decades we have witnessed significant improvements in the possibilities and the ease of interaction with multimedia material. Gradually, the technical and commercial hurdles were overcome to put the user more in control of what, where, and when to enjoy. This development is progressing along different routes. First of all, the availability of digital audio and video compression algorithms, real-time hardware implementations, and associated standards allow efficient storage and transmission of high-quality audio/video (AV) content. In addition, the recent upgrade of access networks, such as cable and telephony networks, provides broadband access to interactive services from the customer premises. As a result, large collections of AV material are becoming accessible via the Internet for on-demand viewing. Another development is that in-home storage of AV material is greatly improving, both in terms of ease of use and storage capacity. VCRs are being replaced on a large scale by storage systems, called personal video recorders (PVRs), based on optical and magnetic or hard disks, and electronic program guides (EPGs) simplify the selection of TV programs for recording. Automatic recording based on recommender technology of interesting broadcast programs has recently been introduced in the market. Modern hard disk technology allows the recording of a considerable number of programs in parallel on a single disk, and playing back a program is possible while its recording is still in progress. The sizes of currently available hard disks allow the storage of hundreds of video files so that they are useful for maintaining a sizable collection at home, thus providing VOD in-the-small.

Current solutions for providing VOD via the Internet rely on best-effort services, meaning that there is no guaranteed level of quality, neither in terms of timely content transmission to allow uninterrupted viewing, nor in terms of (interactive) response times. The protocols used for the transmission of data over the Internet, that is, the Internet protocol (IP) and the transmission control protocol (TCP) provide guaranteed data delivery, but without real-time guarantees. Also the real-time transport protocol (RTP) does not provide any real-time guarantees.

The existing hardware overkill and the possibility to adapt compression and transmission rates to the available bandwidth alleviates this problem, but a temporarily congested network may cause service disruption, or at least deterioration, which may take the form of, e.g., hickups in the display of the video or a slow response to user actions. When the take-up of such a bandwidth intensive service increases, these situations will become unavoidable and appropriate resource management tools such as admission control and reservation protocols as well as scheduling algorithms are required to provide a guaranteed quality of service. The successor of the current IP/TCP protocol suite, called IP Version 6 (IPV6), does provide the means to provide a guaranteed quality of service. Furthermore, an access network like a cable access network provides a controllable environment wherein such protocols and algorithms can indeed be implemented. By placing a VOD server at a central point in the access network, it becomes possible to provide the service with the appropriate quality.

II. CABLE ACCESS NETWORKS

During the nineties of the last century, the realization of the information superhighway dictated the need to connect the homes to backbone networks via broadband links. This problem was known as the last-mile problem. The already existing telephone and cable access networks provided the required infrastructure only partly. Hence, possible implementations, applications, and migration scenarios for these networks were surveyed; see Bisdikian, Maruyama, Seidman & Serpranos (1996),[12]. . Cable access networks, better known as CATV (community antenna television) networks [15], were originally designed for broadcasting analogue video signals. These networks had a tree structure where, at the root, a controller called the head-end (HE) broadcast incoming TV signals from a variety of sources over the network on downstream channels to the individual homes, amplified along the way to retain sufficient signal strength. Over the past fifteen years, CATV networks have been upgraded to provide two-way broadband communication. This upgrading includes replacing parts of the coaxial cable near the HE by fiber-optic cable, organized in rings, extending the functionality of the HE, and installing return amplifiers in the upstream path from the homes to the HE. Access to such a hybrid fiber-coax (HFC) network at the homes requires a cable modem (CM), which separates this public network from the in-home, private networks and provides the necessary functionality for the support of these services.

To ensure interoperability between the HE and a possibly multi-vendor set of CMs, several standards have become available, of which the DOCSIS [MCNS Holdings, 1999] and DVB/DAVIC [Digital video broadcasting, 1999] standards are the two most prominent ones. A third standardization body, the IEEE 802.14 working group, was dismantled in 2000. The drafts remain accessible in their archive [IEEE 802.14 working group, 2000]. These standards describe in great detail the physical (PHY) and medium access control (MAC) layers, covering the electrical characteristics, modulation and error-correction schemes, the message formats and messaging protocols, access protocols and, for DOCSIS in particular, a multitude of quality of- service classes for the support of more advanced services, such as constant bit-rate and real-time polling services. In these standards, there is ample freedom in the design and operation of a system to optimize performance, subject to channel impairments and higher-layer protocol requirements. A HE supports a number of frequency-separated downstream channels with a bandwidth of up to 60 Mbit/s each, where 'M' stands for 220. To each downstream channel, a number of upstream channels are associated, each with a bit rate between 256 kbit/s and 20 Mbit/s. Frequency separation as well as time-division-multiple-access (TDMA) and code-division-multiple-access (CDMA) are used in the upstream direction. In TDMA, the transmission of packets must be separated in time to prevent collisions among them, and in CDMA, multiple packets may be transmitted simultaneously without colliding as long as they use different codes of encoding their packets. Frequency separation is also called FDMA. Besides transmitting the legacy analogue TV signals and, in some countries, digital TV signals, these networks are nowadays predominantly used for Internet-based services via the world-wide web. The number and variety of services are steadily increasing, including communication services such as IP telephony, e-mailing and chatting, search engines for searching information on the web, on-line shops, information services, e.g. on-line newspapers or journals, entertainment such as on-line gaming and video-on-demand (VOD), e-commerce such as on-line booking or banking services, etc. to satisfy all these kind of high-end applications a contention channel is need to be modeled and for that a appropriate queuing is applied to have congestion free environment.

III. THE CONTENTION CHANNEL MODELING

A cable modem (CM) goes through after powering up, where it is trying to get itself registered at the head end (HE). During this phase, it has to search for appropriate channels to receive and send information and go through a series of administrative tasks, such as informing the HE of its capabilities, obtaining operational parameter settings and establishing a first connection. In particular, the CM has to establish tight synchronization with the head end (HE) and set a proper transmission-power level. This part goes by the name of *ranging* CMs are connected to the HE via a tree network of coaxial and fiber cables with amplifiers at the appropriate places in the network. Each CM has its own signal propagation delay to the HE and its own signal attenuation, resulting from its specific location in the network. For efficient operation it is required (*i*) that all CMs are tightly synchronized to allow the transmission of short bursts of information by different CMs to be performed without unnecessarily large, unused time intervals, called guard bands, between successive bursts, and (*ii*) that each CM has a specific power level at which it does its upstream transmissions to the HE to achieve a near-constant reception power at the HE from all CMs.

A CM obtains the information required for ranging from the HE, but this information can only be obtained after a first contact has been established between the CM and the HE, initiated by the CM. For a DVB/DAVIC-compliant system, this initial contact is achieved using a contention-based access protocol, similar to frame-based ALOHA In a contention-based access protocol, multiple messages may be sent simultaneously, resulting in a collision and the loss of all of these messages. A retransmission scheme is employed to ensure that the messages will eventually arrive successfully. In frame-based ALOHA, the slotted time axis is divided into variable-length, consecutive frames. An unsuccessful transmission, in an arbitrarily chosen slot in a frame, can only be repeated in the next one. The central problem in frame-based ALOHA is the computation of the optimal frame length, which clearly depends on the number of contending CMs: a frame that is too small will result in too many collisions, whereas a frame that is too large will result in too many empty slots

IV. CONTENTION ACCESS IN DVB/DAVIC DURING START-UP

The contention procedure during start-up in a DVB/DAVIC-compliant HFC system can be described as follows. The time axis is divided into fixed-length slots. The HE dynamically partitions this slotted time axis into variable-length, nonoverlapping frames. A frame counts an integer number of slots and is aligned with slot boundaries. A frame starts immediately upon reception of a sign-on request message, sent by the HE, which contains the length of this frame. During a frame, each contending CM transmits a sign-on response message to the HE in a randomly chosen slot, uniformly distributed in the frame. This transmission is performed at a particular power level that is under control of a separate procedure. In short, a CM cycles around a number of power levels.

A transmitting CM is either a newly arriving CM that entered its contention procedure during the previous frame and will transmit for the first time in the present frame, or a CM that transmitted in some earlier frame and discovered that its transmission was unsuccessful. A transmission may be unsuccessful because it was done at an improper power level and/or because it collided with a transmission by another CM. A CM discovers that its transmission was unsuccessful by using a time-out mechanism: if a CM does not receive a response from the HE of successful transmission within a maximum feedback delay *T*fb(transmission feedback) after transmission, it considers this transmission unsuccessful. The CM then retransmits in the next available frame, designated by the reception of the first sign-on request message from the HE after this time-out. In practice, this maximum feedback delay is considerable. It is noted that a CM may have skipped various frames before transmitting again in case the frame length is short in comparison with the maximum feedback delay. The main problem is how to determine the frame lengths online so as to make optimal use of the contention channel.

Because unregistered CMs still have to synchronize to the time axis, the slots used by unregistered CMs are made large enough, that is, three times the length of a normal slot. A transmission by a CM close to the HE will arrive (correctly received or not) at the HE at the start of a slot, whereas a transmission by a CM at a larger distance, with a maximum of 80 km according to the DVB/DAVIC standard, will arrive later during the slot. As a result, two transmissions in a single slot need not even collide, and more than one success per slot is possible. In addition, a CM does its successive transmissions at varying power levels, subject to a separate procedure. This generally influences the reception behavior by the HE as well: A transmission may go unnoticed by the HE.

V. QUEUING THEORY AND SYSTEM ANALYSIS

In general people have tendency not to wait for anything. But reduction of the waiting time usually requires extra advancements and investments in the technology. To decide whether or not to invest, it is important to know the effect of investment and advancement on the waiting time. So we need models and techniques to analyze such situations. In this scenario we treat a number of elementary queuing models. Attention is paid to methods for the analysis of these models, and also to applications of queuing models. Important application areas of queuing models are production systems, transportation and stocking systems, communication systems and information processing systems. Queuing models are particularly useful for the design of these system in terms of layout, capacities and control.

A queue is a waiting line. Queuing theory is the mathematical theory of waiting lines. More generally, queuing theory is concerned with the mathematical modeling and analysis of systems that provide service to random demands. A queuing model is an abstract description of such a system. Typically, a queuing model represents (1) the system's physical configuration, by specifying the number and arrangement of the servers, which provide service to the customers, and (2) the stochastic (that is, probabilistic or statistical) nature of the demands, by specifying the variability in the arrival process and in the service process. For example, in the context of computer communications, a communications channel might be a server and the messages the customers; the (random) times at which messages request the use of the channel would be the arrival process, and the (random) lengths of service time that the messages hold the channel while being transmitted would constitute the service process. Another example is a computer system where a programmer (customer) sitting at a terminal requests access to a CPU (server) for the processing of a transaction; both the arrival time of the request for access and the amount of processing time requested are random. Then, the mathematical analysis of the models would yield formulas that presumably relate the physical and stochastic parameters to certain performance measures, such as average waiting time, server utilization, throughput, probability of buffer overflow, etc. The art of applied queuing theory is to construct a model that is simple enough so that it yields to mathematical analysis, yet contains sufficient detail so that its performance measures reflect the behavior of the real system. Queuing theory deals with analyzing congestion problems. Congestion may occur when users share a service system with limited capacity. Whenever the total demand to a system is more than its service capacity, the users should form some queue or waiting line. Users often decide individually when they need a certain service. Due to this uncontrolled arrival process to the system and the often varying service requirements of the users, queues may build up and dissolve over time, which leads to the formulation of stochastic models.

In the context of the cable access network, the upstream channel is the service system and the amount of data that users want to transmit is the service demand. The capacity of the upstream channel is limited, so queues will be formed. The available capacity and the way in which the users are served fully describe the service system. How the queues evolve, though, depends on both the service system and the behavior of the users. Queues cause delay, which for some services could be problematic. That is, delay causes longer transmission times of data and therefore affects the quality of service provided to the user. Consequently, delay characteristics provide measures for the quality of the service system. The upstream channel of cable access networks regulated by a request-grant mechanism might be viewed as a two-stage tandem queue. When a user wants to transmit data, it first joins the request queue where it waits until its request gets granted. Once granted, the user moves to the data queue and waits until its data gets transmitted. The data queue is virtual in the sense that packets are not actually lined up in a queue. Instead, the users hold their packets until they are allowed to actually transmit these. The total service capacity for both queues is equal to the capacity of the upstream channel. How the upstream capacity is scheduled, i.e. divided among the two queues, will determine the delay experienced by the users at each of the two stages.

VI. FEATURES OF QUEUING MODELS

The following properties characterize the kinds of networks that are considered and the behavior that can be studied by queuing theory.

- Flow balance-the number of job arrivals to the network during a time interval equals the number of jobs serviced by the network during the interval.Homogeneity-routing and servicing of a job at a node is independent of the local queue lengths and the mean time between service completions at a given node must not depend on the queue lengths of other nodes. This assumption appears shaky because a packet passing from one queue to another certainly affects the latter strongly because a packet has a fixed length and requires a fixed (and predetermined) amount of service when it enters a queue. Nonetheless, homogeneity is observed in practice. Because gateways service several links in the network and must respond to the traffic of these queues, its response usually is randomized. This gives the assumption operational validity.
- Fair share service-at a node service to a packet is either on a first-come, first-served basis or at random. The same is assumed true within a priority class in a network with access based upon a nonpreemptive, class-priority scheme and service is first-come first-served within a priority class**.**
- Sufficiency of resources41 nodes have infinite buffers and all resources necessary for service are instantly available. There are neither model effects for bottlenecks at a node nor is packet retransmission ever necessary.
- Poisson amvals-the interarrival times of jobs are random and have exponential distribution. For these models, this is a standard assumption.
- Persistence-every node that has **a** packet to transmit is persistent in its effort to acquire service and transmits as soon as service is available (some arbitration scheme is required to break ties).
- Loopless flow-no job requires service at a node more than once.
- Open network- once a job has arrived at its destination, it disappears from the network.
- Reliability-hardware never fails. These effects are not part of the model

No assumptions about the service time distribution have been made, nor are they necessary; queuing theory is sufficiently developed to handle general service distributions. Because the network is open and loopless, the end-to-end delay for a packet is simply the sum of the individual delays in each sub network and the various gateway systems. In other words, the network decomposes completely, allowing us to simply model pieces in relative isolation. The only effect of one piece of the network upon another is the induced traffic load. Predicting the behavior of a network requires knowledge of the traffic load. The size distribution of packets, the rate of packet transmission, and the service time required to send a packet are sufficient to model the average behavior of the network under the given traffic load. Once the isolated behavior of the individual links in the network and the flow of traffic through the network are known, the transmission delay between two nodes for a file many packets long can be calculated.

VII. SCHEDULING AND RESOURCE MANAGEMENT IN CABLE ACCESS NETWORKS

The basic method to transmit data on the upstream channel is by way of a request-grant procedure. If a CM has some data to transmit upstream on behalf of one of the connections it sustains, it first transmits a request to the HE. This request contains an identifier for the connection and an indication of the amount of time, say in terms of a number of slots, it requires to transmit the data. Upon reception by the HE of a request from a CM, it reserves a number of slots and informs the CM about this by transmitting a grant downstream to this CM, indicating that the HE grants exclusive access by this CM to the reserved slots.

Requests are transmitted in dedicated slots, called contention slots, wherein multiple CMs may attempt to transmit a request simultaneously. If this happens, the requests are said to collide and are all lost in the sense that the HE does not receive any of them. To resolve contention, that is, these collisions, a contention resolution protocol is employed that governs the retransmission of collided requests. Hence, at the cost of transmitting relatively short requests in contention, the actual data is transmitted contention-free. Depending on the standard, an alternative to transmitting a request in contention is to use piggybacking, whereby a new request is appended to the data, so that this request is also transmitted contention-free. This, of course, is only possible if the CM has at least one reserved slot at its disposal.

The area of contention-based access to shared media has been an active area of research for decades [Bertsekas & Gallager, 1992; Tanenbaum, 2003]. The type of collision resolution protocol used in a CATV network depends on the standard, and in fact, each of the standards offer several alternatives. One of the main collision resolution protocols employed is based on the well-known ALOHA protocol [Abramson, 1970; Roberts, 1975], the other main protocol is based on contention trees [Capetanakis, 1979; Tsybakov & Mikhailov, 1978; Janssen & de Jong, 2000].

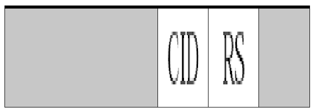
A central problem for the request-grant procedure is how to divide the upstream transmission time into contention and reservation slots to optimize the delay that data incurs. Early work, specifically during standardization, primarily concentrated on extensive simulations, see, e.g., Golmie, Santillan & Su [1999], Sala [1998], Kwaaitaal [1999], Pronk & De Jong [1998], and Pronk, Hekstra-Nowacka, Tolhuizen & Denteneer [1999]. More recently, these simulation experiments have been complemented with analytical results. Palmowski, Schlegel & Boxma [2003], Denteneer [2005] and Van Leeuwaarden [2005] develop queuing-theoretic models for studying the transmission delay in the upstream channel. The latter two are dissertations and contain many useful links to related work.

For reservation-based access in the upstream direction as well as for multiplexing data for connections in the downstream direction, fair queuing algorithms, originally designed for use in switches and routers, play an important role. A fair queuing algorithm aims to guarantee for each connection its fair share of the channel, where the definition of fair share is based on a fluid-flow server that can serve all connections in parallel. These algorithms have for nearly two decades received considerable attention in the literature; see [2] [7] [11] [14]. Besides the basic request-grant mechanism, alternative access modes exist in the upstream direction, such as for providing services with a guaranteed 1.3 Scheduling and resource management 7 quality level. Hence, the HE must generally multiplex more than two access modes on one channel [Pronk, 2000].

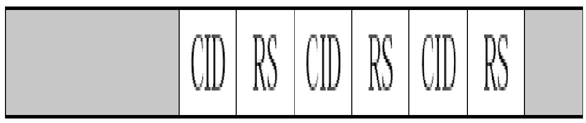
A downstream channel is typically coupled to four or eight upstream channels, and CMs may switch between these upstream channels, as well as switch between downstream channels and, consequently, upstream channels. These migrations are under control of the HE, and leads to the problem of load balancing among the channels. The time-varying behavior of a CM in terms of its load on the network, both downstream and upstream, require load balancing algorithms to operate on-line.

VIII. DEFINING MULTI-REQUESTS

A request frame carrying a single request is typically composed of a header, and a connection ID and request size field. This header includes (*i*) physical layer overhead such as a guard band for safely separating subsequent bursts from different CMs and a preamble that is required for demodulation of the received burst, and (*ii*) some medium-access-control-layer overhead, such as a control field for identifying the request as such and a check sequence for error detection. As a result, only a relatively small part, typically one third, of the entire request is dedicated to the connection ID and the request size. Figure 3.1a illustrates the structure of a single request frame. Defining a multi-request frame can be done by extending the request frame with more (connection ID, request size)-pairs, resulting in a relatively modest increase in total size per added pair, see Figure 1. In a practical situation, it is reasonable to assume that a multi-request can contain up to a maximum number of requests. Suppose that the length of a request frame is 1, and that a multi-request can consist of a maximum of *R* requests, then the length of a multi-request frame can be expressed as $1+\alpha(R-1)$, with $0 \leq \alpha < 1$. This maximum *R* can be made dependent on the current load, which can be estimated from the number of requests received per multi-request. The successful reception by the HE of a multi-request is equivalent to the successful reception of all of the requests it accommodates. Identically to single requests, the size of a request in a multi-request can be updated to incorporate new data. In addition, if for a CM, its current multirequest is not yet full, that is, it carries less requests than it can maximally contain, a request from a newly active connection can be added on the fly to the multi-request by including it in its next transmission. After all, also a multi-request may have to be transmitted repeatedly before it is successfully received by the HE.



(a) A single request frame



(b) A multi request frame

Figure 1. Structure of a single-request and a multi-request frame.

The grey areas are physical and medium-access control-layer overhead, CID stands for connection ID, and RS for request size.

IX RESULTS AND OBSERVATIONS

To achieve congestion avoidance environment and to have reliable communication in cable access network an environment is created and the performance parameters are tested on this environment. It's clearly observed from the following plots that by applying queuing models we can achieve all the performance criterions like transmission delay, goodput, blockage probability and number of hops taken by a reliable path.

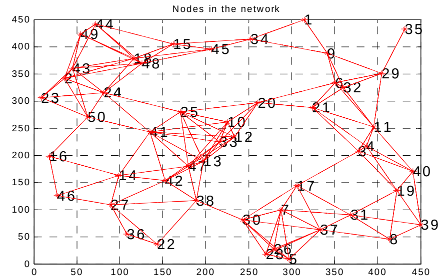


Figure 2: Nodes in the network

The above plot indicates that the network environment created for testing. It is clearly seen that we are testing on 40nodes and there is provision provided that an individual can select what is source and destination on his own and can check for any source and destination pair. The main goal will be the data packets which we are going to transfer in this network is been sent with out any congestion or not. The path which it chooses should be reliable path which is done by M/M/1 and M/G/1 queuing models. In our approach they are named as static and dynamic environments. The source and destination which we have chosen are low resource route and hence the path couldn't be established. Hence we are giving option for another source destination pair. If we accept (1) then another route will be established. Upto when the route gives the reliable and congestion free path this procedure executes.

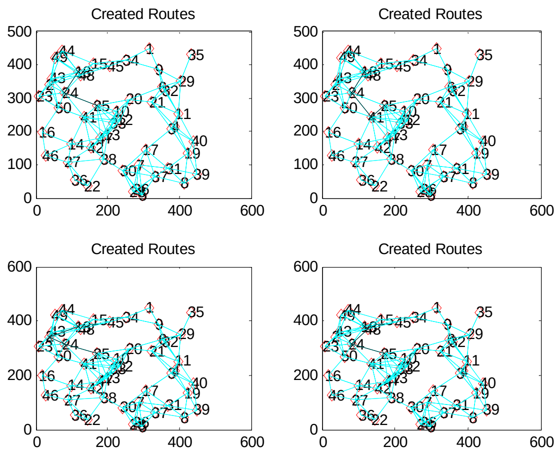


Figure 3: possible routes(on applying queuing models)

The above plot indicates all the possible routes for transferring the data packets without any delay and congestion. This will be taken care by the queuing model. Why because the nodes will be in the mobile state. Once the node from their position also we need to deliver the packets. This is analyzed on nodes Markovian behavior (random). The above source destination pair didn't give reliable path but the above plot can be observed that the 38-26(s/d) link is giving us result that the destination is reached and it is also mentioned that how many number of hops it has taken. Because the number of hops indicate the delivery rate.

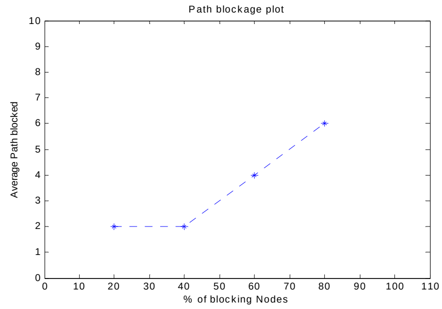


Figure 4: Path Blockage Plot

The above is plot between percentage of blocking nodes vs average path blocked. Once the data packets starts travelling in the cable access networks it should be clearly under observationn that how many nodes are blocked during heavy traffic. Once the nodes are blocked the data packets wont reach on delivery time and there is a chance of data lossage. It clearly mentions that during the journey of datapackets on applying queuing models how the blockage probability is and how it checks for the route which gives next route.

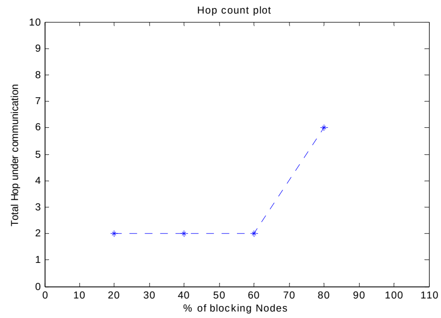


Figure 5: Hop count Plot

Hopping means nothing but how source node delivers its packets to relay nodes, gateway nodes and from those nodes how they deliver their packets to the destination node. Hopping provides a cooperative communication between participated nodes. Its clearly known that less number of hops good delivery rate.

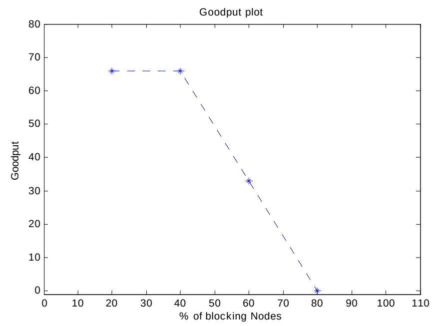


Figure 6: Good put

In computer networks, goodput is the application level throughput, i.e. the number of useful information bits, delivered by the network to a certain destination, per unit of time. The amount of data considered excludes protocol overhead bits as well as retransmitted data packets. This is related to the amount of time from the first bit of the first packet is sent (or delivered) until the last bit of the last packet is delivered. By applying our methodology when any routes are blocked how it tackles the situation and delivers the packets in reliable route are shown which improves the good put of the system.

X. CONCLUSION

It is important to prevent access network from bottlenecking the end-to-end survivability, especially as voice, video, and data traffic are all delivered through the same access network. With economic constraints and limited routing capability, the structure of an access network is typically a challenging design, where the terminal has to relay the traffic from another terminal of the same or higher level. The impact of variations in traffic patterns and intensities, channel capacities, and message protocols can be assessed using them because of the lack of fine detail in the network traffic rates, traffic patterns, and the hardware used to implement the networks. We designed a access network which is congestion free and it is also observed that by the help of queuing models a best effort traffic is achieved. Good scheduling, reliability and congestion free environment is been observed on static as well as random networks.

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