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IMPROVING THE PERFORMANCE OF INTEGRATED SERVICE NETWORKS USING SELF ADAPTIVE CSMA/CD AND VARIOUS CONTROL ALGORITHMS

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Abstract- In this paper, a study of mixed voice and data on a local area network using Carrier Sense Multiple Access with Collision Detection (CSMA/CD) is presented. In multi-access/broadcast systems, each station is connected to a common communication medium (such as a bus) through an interface to listen to all transmissions and copy packets that are addressed to it. Since no more than one transmission can be carried on the bus at a time, the stations have to share the bus by means of a multiple access protocol. A variation on the carrier sense multiple accesses with collision detection (CSMA/CD) protocol for local area random access broadcast network was presented for supporting the facility of mixing both voice and data traffic over the bus LAN. This protocol transmits into an idle channel with a dynamic probability (P) and back of with probability (1-P). To handle the collision state, we examine various retransmission policies. Simulation results are presented to illustrate the effectiveness of the various retransmission approaches in comparison with the traditional CSMA/CD that uses binary backoff algorithm to handle the retransmitted packets in case of existing a collision in the channel. It is concluded that by suitable choice of the control parameter and the appropriate retransmission policy, optimum throughput can be achieved and the voice packet loss due excessive delay may be kept below the threshold of 0.52 percent.

Keywords: AP-CSMA/CD, Linear Backoff, Fixed Backoff, Exponential Backoff, Decoder, Integrated Service Digital Networks, Ethernet, Multiple Accesses.

I. INTRODUCTION

A network consists of multiple computers connected using some type of interface, each having one or more interface devices such as a Network Interface Card (NIC) and/or a serial device for PPP networking. Each computer is supported by network software that provides the server or client functionality. The hardware used to transmit data across the network is called the media. It may include copper cable, fiber optic, or wireless transmission. The standard cabling used for the purposes of this document is 10Base-T category 5 Ethernet cable. This is twisted copper cabling which appears at the surface to look similar to TV coaxial cable. It is terminated on each end by a connector that looks much like a phone connector. Its maximum segment length is 100 meters. There are two main types of network categories which are: Server based and Peer to Peer based. In a server based network, there are computers set up to be primary providers of services such as file service or mail service. The computers providing the service are called servers and the computers that request and use the service are called client computers.

In a peer-to-peer network, various computers on the network can act both as clients and servers. For instance, many Microsoft Windows based computers will allow file and print sharing. These computers can act both as a client and a server and are also referred to as peers. Many networks are combination peer-to-peer and server based networks. The network operating system uses a network data protocol to communicate on the network to other computers. The network operating system supports the applications on that computer. A Network Operating System (NOS) includes Windows NT, Novell Netware, Linux, UNIX and others.

Another type of network called Integrated Services Digital Network (ISDN), it is comprised of digital telephony and data-transport services offered by regional telephone carriers. ISDN involves the digitization of the telephone network, which permits voice, data, text, graphics, music, video, and other source material to be transmitted over existing telephone wires. The emergence of ISDN represents an effort to standardize subscriber services, user/network interfaces, and network and internetwork capabilities. ISDN applications include high-speed image applications (such as Group IV facsimile), additional telephone lines in homes to serve the telecommuting industry, high-speed file transfer, and videoconferencing.

Voice service is also an application for ISDN. Many papers have been written on the topic of integrated voice and data on LANs including CSMA/CD. Most of these studies [1,6] have shown that CSMA/CD LAN performance tends to degrade rapidly as the load exceed about 40 % of bus capacity and it is reaffirmed in the paper [2] for 1 Km, 10 Mbps bus. Above this load, the number of packet collisions rises rapidly due to the interaction among repeated transmission and the new packet arrivals. Collided packets backoff using various retransmission policies described in this paper.

In [3], Gonsalves has simulated the performance of the CSMA/CD protocol. It was shown that the correct choice of the back-off delay would minimize the number of collisions and maximize the throughput through the LAN. The optimum mean Backoff delay depends on the number of stations connected across the network. This choice is difficult to make as the number of active users over a period of time varies randomly. In this paper, we use an adaptive P-CSMA/CD protocol (AP- CSMA/CD) which optimizes the success possibility of packet transmitted and yet facilitates voice to be integrated into a bus network [2]. To fulfil the conflicting demands of voice and data traffic. the voice traffic is allocated higher priority. We have included in this paper various back-off algorithms instead of an intrinsic back-off capability [4] into the voice and data stations to optimize the network perfom1ance.

The output results of the simulation show that both voice and data traffic can be supported over the bus with guaranteed voice quality but differ from one policy of retransmission to other according to various network parameters. To verify the above objective, we present this paper which is organized from the following sections: section two describes the applied protocol. In section three, we discuss the various policies used to handle the collision state. Section four was dedicated for the perfom1ance measure. In section five, we present the simulation results. Finally, the paper was enclosed in section six with conclusions and further development to be done by others.

II. NETWORK TOPOLOGIES

In general, the computer network topologies are classified into the following types:

A. Bus Topology

Both ends of the network must be terminated with a terminator. A barrel connector can be used to extend it. Bus topology uses a common backbone to connect all the network devices in a network in a linear shape. A single cable functions as the shared communication medium for all the devices attached with this cable with an interface connector. The device, which wants to communicate send the broadcast message to all the devices attached with the shared cable but only the intended recipient actually accepts and process that message.

Ethernet bus topologies are easy to install and don't require much cabling and only a main shared cable is used for network communication. 10Base-2 and 10BaseT are two popular types of the Ethernet cables used in the Bus topology. Also, Bus network works with very limited devices. Performance issues are likely to occur in the Bus topology if more than 12-15 computers are added in a Bus Network. Additionally, if the Backbone cable fails then all network becomes useless and no communication fails among all the computers. Unlike in the Star topology in which if one computer is detached from a network then there is not effect on the other computers in a network.

B. Ring Topology

In ring Network, every computer or devices has two adjacent neighbors for communication. In a ring network, all the communication messages travel in the same directory whether clockwise or anti clockwise. Any damage of the cable of any cable or device can result in the breakdown of the whole network. Ring topology now has become almost obsolete. FDDI, SONET or Token Ring Technology can be used to implement Ring Technology. Ring topologies can be found in office, school or small buildings.

C. Star Topology

In the computer networking world the most commonly used topology in LAN is the star topology. Star topologies can be implemented in home, offices or even in a building. All the computers in the star topologies are connected to central devices like hub, switch or router. The functionality of all these devices is different. I have covered the detail of each networking devices in the separate portion of my website. Computers in a network are usually connected with the hub, switch or router with the Unshielded Twisted Pair (UTP) or Shielded Twisted Pair Cables. As compared to the bus topology, a star network requires more devices & cables to complete a network. The failure of each node or cable in a star network, won't take down the entire network as compared to the Bus topology. However if the central connecting devices such as hub, switch or router fails due to any reason, then ultimately all the network can come down or collapse.

D. Tree Topology

Tree topologies are comprised of the multiple star topologies on a bus. Tree topologies integrate multiple star topologies together onto a bus. Only the hub devices can connect directly with the tree bus and each Hub functions as a root of a tree of the network devices. This bus/star/hybrid combination supports future expandability of the computer networks, much better than a bus or star.

E. Mesh Topology

Mesh topology work on the concept of routes. In Mesh topology, message sent to the destination can take any possible shortest, easiest route to reach its destination. In the previous topologies star and bus, messages are usually broadcasted to every computer, especially in bus topology. Similarly in the Ring topology message can travel in only one direction i.e. clockwise or anticlockwise. Internet employs the Mesh topology and the message finds its route for its destination. Router works in find the routes for the messages and in reaching them to their destinations. The topology in which every device connects to every other device is called a full Mesh topology unlike in the partial mesh in which every device is indirectly connected to the other devices.

There are also hybrid networks including a star-bus hybrid, star-ring network, and mesh networks with connections between various computers on the network. Mesh networks ideally allow each computer to have a direct connection to each of the other computers. The topology this documentation deals with most is star topology since that is what Ethernet networks use.

III. DESCRIPTION OF THE AP- CSMA/CD PROTOCOL

The principle of CSMA/CD (Carrier-Sense Multiple Access w. Collision Detection) is that several stations sharing a common communication medium may access this medium instantly if they want and the medium is presently idle. The Carrier-Sense capability of the stations insures that no running transmissions are disturbed; however, carrier cannot be detected before a certain signal propagation delay. Therefore some stations may 'collide' in the attempt to transmit.

By comparing their outgoing signal to the incoming on the medium, stations are able to detect this; they stop transmitting (back off) and retry after some time. It is reasonable to choose this time to be a random multiple of the slot-time, which is defined as to exceed the worst-case signal round-trip delay on the medium. In the standard CSMA/CD protocol, each active station senses the channel and transmits a packet with probability one (I-persistent CSMA/CD) if the channel is sensed free. This works extremely well in light load conditions. In the event that two or more station sense a free channel together, they will transmit together into the channel and cause packet collisions. A jamming signal will be sent into the channel by these active stations involved in the collisions. In IEEE 802.3 standard, each station will apply the truncated binary Backoff algorithm and reduce the possibility of repeated collisions of these packets. Under light load, the algorithm works well for voice or data transmission. However, under heavy traffic, the accumulated delay of the voice packet due to repeated collisions of the transmitted packet render this protocol inadequate for carrying voice and data.

In the used AP-CSMA/CD protocol [4], the probability of transmission when a station senses a free channel is controlled by two parameters: P_v the probability of voice packet transmission and P_d the probability of data packet transmission. Upon sensing a free channel, a station using AP-CSMA/CD protocol will transmit a ready packet with probability P_v for voice packet or P_d for data packet. A station, which did not transmit, will wait for a minimum period t_p (end-to-end propagation delay of the bus) before it reattempt its transmission. This algorithm eliminates the probability of collision due to reattempt by the same station during a vulnerable period of transmission of other stations. In the event of packet collisions, all active stations involved in the collision will send jamming signal into the channel. Stations involved in the collision will attempt transmission after the channel is free again. Here, there are simple dynamic probabilities of transmission used to transmit voice and data denoted by P_v and P_d. In order to support voice and data traffic over a bus, an algorithm that assigns values to dynamic probabilities P_v and P_d is formulated [4]. This algorithm assigns different priority to voice over data. Under light load, the initial values P_{vi} and P_{di} are assigned to P_v and P_d to reduce delay and both of these probability decreases monotonically to final values P_{vf} and P_{df} under heavy loading conditions.

IV. VARIOUS POLICIES FOR HANDLING COLLISION STATE

A collision occurs when two workstations on the network both sense that the network is idle and both start to send data at approximately the same time, resulting in a garbled transmission. The term collision itself seems to imply that something is wrong. In some technical literature, this kind of event is called a stochastic arbitration event, or SAE, which sounds much less like an error than does collision. However, collisions are expected in older Ethernet networks. Only when they become excessive is it time to search for the sources of the collisions and rearrange some workstations or network devices as appropriate. The collision domain has pretty much been relegated to history. Hubs and half-duplex connections still use CSMA/CD, but if your network uses Fast Ethernet switches, in full-duplex modes, then CSMA/CD no longer come into play. Instead, full-duplex switches use separate wire pairs in the cable so that the switch port can send data to the attached computer, while receiving data from that computer on another wire pair. When creating a new network today, the cost of network adapters and switches makes it a very inexpensive proposition to use full-duplex network adapter cards and switches. The CSMA/CD technology is discussed in this chapter to let you understand how Ethernet has evolved, and to provide information for those who still have legacy Ethernet equipment installed.

Make absolutely sure that when you plan for your network ports you are using technology that supports the settings you intend to use. Older systems (NICs, switches, and so on) use older technology that cannot auto-negotiate the settings properly, thus causing slowdowns and other issues. Make sure that when in doubt, set both sides of the network connection for auto-negotiation, or only set them to the exact standards in which they can operate. Without a backoff algorithm, the device that detects a collision will stop and then try once again to transmit its data onto the network. If a collision occurs because two stations are trying to transmit at about the same time, they might continue to cause collisions because both will pause and then start transmitting at the same time again. This will occur unless a backoff algorithm is used. The backoff algorithm is an essential component of CSMA/CD. Instead of waiting for a set amount of time when a device backs off and stops transmitting, a random value is calculated and is used to set the amount of time for which the device delays transmission [13].

The calculation used to determine this time value is called the Truncated Binary Exponential Backoff

Algorithm. Each time a collision occurs for an attempted transmission for a particular frame, the device pauses for an amount of time that increases with each collision. The device tries up to 16 times to transmit the data. If it finds that it cannot put the information onto the network medium after 16 attempts, it drops the frame and notifies a higher-level component in the protocol stack, which is responsible for either retrying the transmission or reporting an error to the user or application. A method similar to CSMA/CD is CSMA/CA, in which the last two letters, CA, stand for collision avoidance. Networks that use this method access the physical medium such as Apple Talkand listen to the network just as an Ethernet device does. However, before sending out a frame on the network, networks using CSMA/CA first send out a small packet indicating to other stations that they are about to transmit. This method helps to greatly reduce collisions but is not widely used because of the overhead produced when its networks send out the informational packet. The IEEE 802.11 wireless networking standard also uses CSMA/CA as its network access method

Algorithms for the control of the retransmission procedure in random multiple access schemes are needed to ensure stability of the system operation under high traffic conditions. Optimal retransmission control policies cannot be applied in practice since they are based on global information about the system state. In case of existing the collision mentioned in the previous section, we do not simple mean use of forward error correction techniques but rather randomized retransmissions of packets or, more generally, transmission of redundant packets. Here, we focus on specific schemes that employ the above philosophy and we sack to establish their performance. We consider three policies; the first one assumes random access, with each packet being retransmitted randomly again and again until a fixed period of Time from time to time of this initial generation clasps (α policy). The second policy is identical to the first one except that the retransmissions continue randomly until a fixed number M of them is compacted (β policy). Assuming the rescheduling times is independent and exponentially distributed with parameter γ and setting $\gamma^*C = \gamma_c$. We can consider an obviously superior third policy that requires the retransmission to continue either until M submissions are completed or until v seconds pass, whichever occurs first (γ policy) [5, 11, 12].

V. OBSERVED PERFORMANCE PARAMETERS

In this section, we say that the main parameters that determine the performance of the network are defined by the following two parameters:

- First: The speech delay, D is defined as follows [4]:

$$D = \text{packetization delay} + \text{queuing delay} + \text{transmission delay} \quad (1)$$

where:

D is the measures of time taken for a speech sample generated at source to be transmitted to the destination buffer ready for reproduction. This delay depends on the instantaneous values of P_v and P_d.

- Second: Throughput of the channel is defined as:

$$S = T_s / T_t \quad (2)$$

where:

S: a measure of the busy period of the channel transmitting packets without collisions to the total Capacity of the channel. This period is high sensitive to the values of P_v and P_d. If optimum values of P_v and P_d are selected, maximum throughput of the channel is guaranteed

T_s: Successful transmission time

T_1: Total simulation time

VI. NETWORK MODEL, SIMULATION PARAMETERS AND RESULTS

In the work presented in this paper, we consider the model described in Figure 1. Also, we propose a simulator to detem1ine the characteristics of broadcast networks. In this simulator, the protocols used by the sources, the traffic generated by a source, the physical position of a source on the channel, the parameters that define a transmission channel and the interaction of channels can be varied. When the simulator runs, time jumps between the occurrence of significant events, rather than continuously moving forward by small increments.

Functions such as wait for idle channel and listen while transmitting are implemented by giving the channel the characteristics of an active device. The channel notifies a source when the static of the channel, at the location of the source, has changed. Actually, the channel in this type of system is passive and sources that perform these functions examine the channel continuously. However, the interval of time in which the static of the channel can change and a significant event can occur is small compared to the average time between significant events [7, 8]. A simulator that models the continuous operation of the sources would have to examine the channel at small intervals, and would be much less efficient, in terms of processing, than the technique selected. In the used simulator, the channel maintains lists of sources that are transmitting, listening while transmitting, and listening for an idle channel.

When the state of the channel changes, channel maintenance routines notify the terminals in the lists at the appropriate times. These routines take into account the propagation delay between terminals, and the detection time of terminals. Queues of massages and packets waiting to be transmitted from the sources are maintained without creating physical queues. Instead, the clock associated with packet and massage generation is allowed to run independent of the simulation clock. Each source keeps track of the time the packet and the message it is transmitting were generated. When the transmission is complete, this time is updated to determine the next packet or message arrival at this source [9, 10]. If this time is less than the simulation time, this object has been waiting in a queue, and can be transmitted immediately. If it is greater than the simulation time, the queue is empty, and the message or packet is scheduled to arrive in the future.

The voice terminals only transmit during active speech intervals. For the parameters selected in the simulations, the speech interval duration's are more than three orders of magnitude greater than the packet transmission time, and more than six orders of magnitude greater than certain of the channel related events. The simulator tracks the fine structure of channel related events. Because of the different time scales, the simulation could not be conducted for a long enough time to guarantee that all combinations of active and inactive intervals occur with the proper probabilities. To compensate for this, a different random number generator is used to generate active and silent intervals than is used to generate the other random events in the simulation. This allows different protocols for voice sources to be compared when the same active and silent intervals occur. The simulated system has the following characteristics [14]:

- Sources are uniformly distributed along a single cable;
- Channel length is 2 km;
- Transmission rate is 10 Mbps;
- Time to detect the presence and removal of carrier T_{on} and T_{off} is 1 μs;
- Time to detect inference is 1 μs;
- Time the channel must remain idle between transl1lissions is one- third of a microsecond;
- Voice packets arc generated by different stations randomly;
- A vectored in each station produces voice samples at 64 kb/s;
- All silent packets are discarded;
- The talk spurt and silence intervals alternate and arc assumed to be exponentially distributed;
- Mean talk spurt and silence intervals arc 0.17 and 0.41 sec;
- Jamming signal for handling packet collision is 32 bit long;
- Number of connected stations equal 1024;
- Max. Number of active voice and stations are 100 each;
- Transmission buffers and receiving buffers store up to 5 packets each;
- Any voice packet. Which is delayed? More than 20 ms in the buffer is discarded and considered as lost packet;
- Data packets are discarded only when the transmission butters overflow;
- The message enters arrival process is a negative exponential distribution with a mean of 33.47 ms;
- The packet size is 1024 bits;
- The message length is deterministic, it is 960 bits;
- The minimum time between successive packets from the same source is four round-trip Transmission delays;

The model of transmission strategy used by the sources in the simulation is shown in Figure 2. In the first phase of simulation, the model with its assumptions described above was examined against the standard CSMA/CD model described in [1]. From Figure 3, we see that the network model which uses AP- CSMA/CD transmission protocol performs better than the standard CSMA/CD protocol. This figure shows the throughput of both protocols as a function of offered load. The throughput attainable by the AP-CSMA/CD protocol with the third policy of retransmission (γ policy) is about 12 percent higher in the overload conditions. For offered load less than 5, both throughputs are nearly equal. Also, AP-CSMA/CD protocol may support sixty percent bus loading before any packet is discarded. This is substantial gain a 50 percent channel capacity over the 40 percent available in the standard CSMA/CD protocol. At low load, the AP-CSMA/CD algorithm estimates the mean number of stations with at least one packet in the queue. This number changes more erratically for 100 stations than for 500 stations as Packets are distributed to each station according to a uniform random number. As the load becomes large, every station on the bus has a packet in the queue and thus allows the optimizing algorithm to steer P to a tight tolerance. Figure 4 shows the throughput performance of the AP- CSMA/CD for 100 voice calls using various types of backoff algorithms. For P_v between 0.02 and 0.035, a high throughput is achievable for 100 voice calls knowing that the (γ policy) backoff performs better than the truncated back off. Outside this range, the throughput drops rapidly in case I relative to case 2 due to excessive delays as a result of packet collisions or unnecessary queuing delay due to the small value of P_v. Figure 5 shows the average delay for increasing P_v.

LBT: listen to the channel before transmitting
LWT : listen to the channel while transmitting
XMIT: transmit
WIC: wait for idle channel
SRB: schedule a retry after a busy channel
SRC: schedule a retry after a collision
SNP: schedule transmission of next packet
SNM: schedule transmission of next message
B: channel busy
B: channel not busy
C: collision during transmission
C: no collision during transmission
LP: test if the transmitted packet was the last packet in the message
LM: test if the transmitted message was the last message in the queue
WNM: wait for next message.

VII. CONCLUSIONS AND FUTURE WORK

In this paper, we have used the P-CSMA/CD protocol for integrating voice and data on a bus network under working with various types of back off algorithms. The AP-CSMA/CD protocol with an optimum retransmission policy facilitates higher the priority be allocated to voice transmission by means of the parameters P_v and P_d. Also, it was shown that the mixed retransmission back off algorithm gives better performance than the BEK algorithm that is used in the stander CSMA/CD protocol. The values of P_v and P_d should be adjusted to optimize the usage of voice and data bus network. We recommend applying the partitioning concepts on the connected stations across the network in order to reduce the effect of collisions that happened as a result of heavy traffic, after that we need to check the output performance of this network type.

At high offered loads the fairness of the Ethernet protocol causes degradation of performance of all voice connections. Thus to ensure acceptable quality some higher-level access control algorithm is necessary to avoid peaks in the offered load. This is especially important in an integrated voice/data network in the absence of packet-level priorities. This is an area for further study. An important contribution of this paper is that we have substantially increased the base of experimental data available on the performance of the 3 Mbps experimental Ethernet under diverse conditions. This database can be used to develop and validate theoretical models for the purpose of studying other implementations of the Ethernet architecture for various applications. This is currently under investigation. Although our results were obtained using a specific voice protocol, we feel that using other protocols would not improve performance appreciably. In fact, since we have not considered software and protocol overhead, our results may represent upper bounds on the performance of the 3 Mbps Ethernet for voice applications.

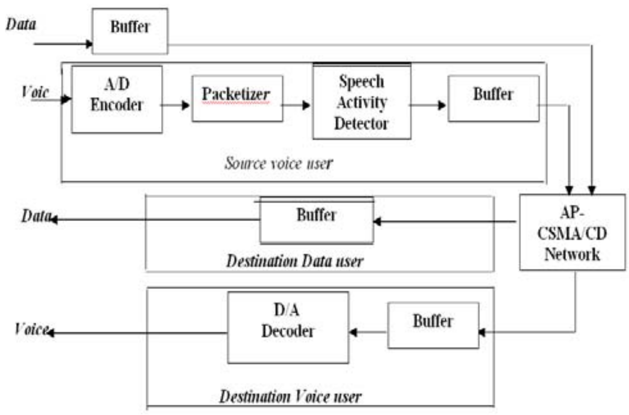


Figure 1. Voice and data model

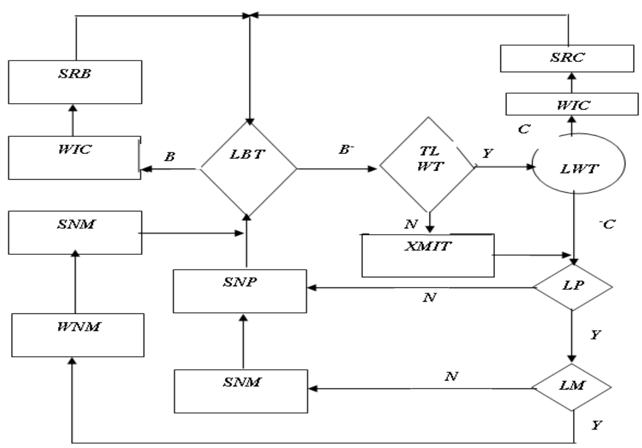
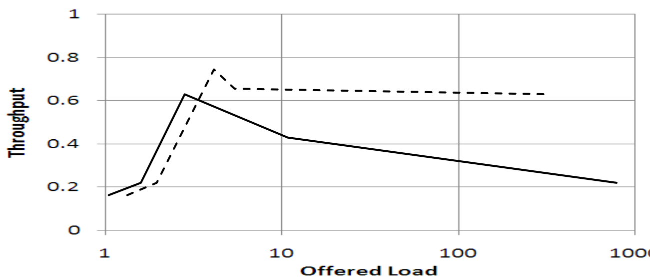
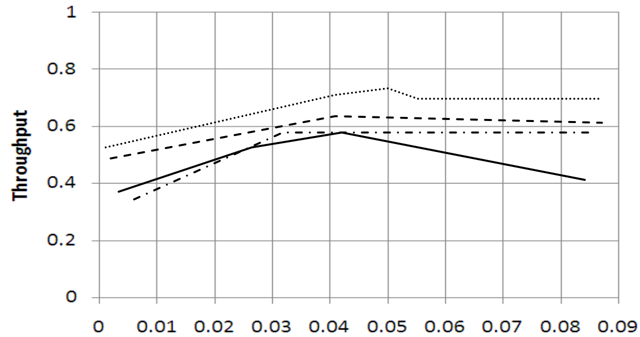


Figure 2. Model of transmission policies used by sources in the simulation process



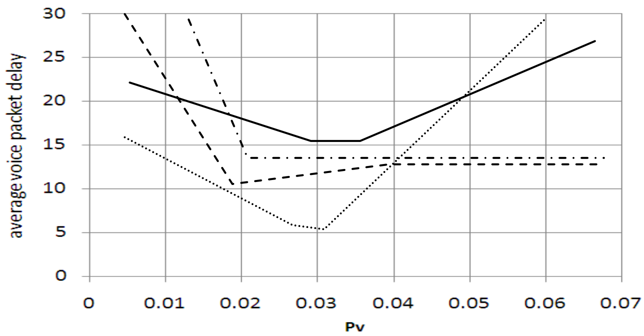
Case 1: - - - - - - - - - - Buffer size= 5 with AP-CSMA/CD
Case 2: -----------------Buffer size= 5 with CSMA/CD

Figure 3. Throughput versus offered load of two types of CSMA/CD protocols



Case 1: ………….. 100 voice calls (γ)
Case 2: - - - - - - - 100 voice calls (α)
Case 3: -.-.-.-.-.-. 100 voice calls (β)
Case 4: ----------- 100 voice calls (BEK policy)

Figure 4. 10 MBPS AP-CSMA/CD: Throughput versus P_v for 100 voice calls using various types of backoff algorithms



Case 1: ---------------- 100 voice calls (BEK policy)
Case 2: …………………... 100 voice calls (γ)
Case 3: -.-.-.-.-.-.-.-.- 100 voice calls (α)
Case 4: - - - - - - - - - - 100 voice calls (β)

Figure 5. 10 MBS AP-CSMA/CD: Av. packet delay versus P_v for 100 voice calls using various types of backoff algorithms

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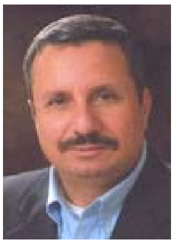
BIOGRAPHIES



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