Real-Time Communication in Packet-Switching Wide-Area Networks

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TR-89-022

May, 1989

Abstract

The increasing importance of distributed multimedia applications and the emergence of user interfaces based on digital audio and digital video will soon require that computer communication networks offer real-time services. This paper argues that the feasibility of providing performance guarantees in a packet-switching wide-area network should be investigated, and describes a possible approach. We present a model of the network to be studied, and discuss its generality, as well as the presumable limits to its validity in the future. We also formulate the problem, give a definition of the guarantees to be provided, and describe a correct scheme for the establishment of real-time connections with deterministic, statistical, and best-effort delay bounds.

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1 Computer Science Division, EECS Department, University of California and the International Computer Science Institute. This research has been supported in part by the Defense Advanced Research Projects Agency (DoD), ARPA Order No. 4871, monitored by Naval Electronic Systems Command under Contract No. N00039-84-C-0089, the University of California with a MICRO Program grant, Cray Research, Inc., Hitachi, Ltd., IBM Corporation, Ing. C. Olivetti & C., S.p.A., and the International Computer Science Institute. The views and conclusions contained in this document are those of the author, and should not be interpreted as representing official policies, either expressed or implied, of any of the sponsoring organizations or of the U.S. Government.

1. Introduction

The current evolutionary trends of computing and networking technologies point to a major increase, over the next several years, in the need for real-time communication, i.e., for computer communication with guaranteed performance. Performance guarantees are required not only in distributed process-control and military systems, but also in supercomputer-to-supercomputer communications and even in such business applications as stock trading. Furthermore, guarantees will be needed by all those systems that process and transmit digital audio, digital video, or any combinations of each with other types of information. Multimedia distributed systems are expected to emerge in the near future, and to become predominant shortly thereafter, as the newer audio- and video-based user interfaces are added to the traditional ones (data, text, and images).

Providing performance guarantees typically means guaranteeing that, given certain bounds on the flow of information into the network, the delay with which each information item reaches the destination will satisfy given bounds. For interactive audio and video, the bound will usually be on the value of the delay of each bit, or byte, or message transmitted by a given sender to a given receiver; for instance, it is well-known that in a phone conversation delays longer than half a second are sufficient to make the talkers uncomfortable. In the case of non-interactive audio and video communication (e.g., the transmission of music or a movie for immediate or deferred consumption by the receiving user), performance indices more important than the actual delay are its variance and, even more, its jitter. Once a network or internetwork offers real-time services, other applications of these services are likely to emerge; for example, it may be desirable to have some remote procedure calls and their replies, and some accesses to remote files or databases, performed within certain maximum time intervals. The availability of real-time services, if their prices are not prohibitive, is likely to expand the demand for them well beyond the boundaries of those applications that are motivating their introduction.

Some of the solutions proposed for the problem of designing network protocols and management policies to support real-time communication apply to local-area networks. They include such media access protocols as the Fiber Distributed Data Interface (FDDI) and the High Speed Ring Bus (HSRB); both of them provide guaranteed delay, FDDI using the timed token scheme [Ross86], and HSRB the reservation/priority mechanism [SAE86]. Circuit switching can be used for this purpose in wide-area networks as well as in LANs: by dedicating a circuit of fixed bandwidth to each connection for the entire duration of the connection we can bound delays. However, circuit switching is not the most convenient technique for data communication [Harr80]. Hence, schemes that combine packet and circuit switching features in various ways have been proposed for integrated voice/data networks: fast circuit switching, burst switching, hybrid switching, and fast packet switching. An overview of these schemes can be found in [Chen88]. The most promising of them is fast packet switching, which is based on a major simplification of the lower layers in the protocol hierarchy; this simplification yields real integration of services (at the lower layers, all packets are treated in the same way, no matter whether they contain data or voice, or perhaps even video) as well as the possibility of protocol implementation in hardware. It is important to notice, however, that fast packet switching can provide low but not necessarily bounded delays, and therefore does not per se offer true real-time communication.

This paper addresses the problem of providing performance guarantees in a wide-area network or internetwork. The solution we present applies to all current and future networks that are adequately represented by the model in Section 2. The formulation of the problem and our approach to its solution are discussed in Section 3. An algorithm that enables a network of the type defined in Section 2 to offer real-time services is described in Section 4. Finally, Section 5 presents our conclusions and plans for future work.

2. The Network

In its full generality, the network we consider in this study has an arbitrary topology, and may actually result from the interconnection of several networks of various types (LANs and WANs). In it, a message can go from a source host to a destination host passing through a number of intermediate nodes, some of which may be gateways (or routers) connecting one network to another. The functions of a network node may be implemented by hosts or by special purpose communication computers or by both. A source-destination path goes through a number of nodes where the transmitted information can be stored and then forwarded to the next node. Some of the links between adjacent store-and-forward nodes may actually be non-store-and-forward networks, provided their total delay can be bounded. They could be, for example, very high-speed circuit-switched trunks, whose transmission speed is so high that there can be no storing or processing in the switches. The traffic on these trunks will generally include packets from different sources to different destinations, but the delay of each packet will be bounded. Thus, the only assumption we make about the links between store-and-forward nodes is that there be a known and finite bound for the link delay of each packet. Without it, we would not be able to offer hard real-time guarantees. It should be noted that this assumption is not exactly satisfied by links governed by contention-based protocols (e.g., Ethernets, see [Kuro84]), but it is by other types of LANs, such as FDDI rings (see for example [Sevc87], [Dyke88]).

We believe that the model presented above, though not absolutely general, is an adequate characterization of many current and future wide-area networks. How long in time will its validity last is an open question: at some point in the future, store-and-forward nodes might disappear altogether, or some of the links might exhibit unboundable delays (even now, the presence of an Ethernet along the path of a real-time packet raises the specter of invalidity when absolute delay bounds are desired). However, it is hard to predict when the number of networks satisfying our model will become negligible.

A network with these properties could be based on circuit switching, or, since circuit switching is a rather wasteful technique in data communications, on hybrid switching. In the latter case, circuits would be allocated to real-time connections, while non-real-time communications could use the remainder of the bandwidth of each link in packet switching mode. If the boundary between the two portions of a link's total bandwidth is made movable as a function of demand, this solution can exploit network resources more efficiently, but requires fairly complex switching in each node. This in turn increases the difficulty of designing the very-high speed switches that the higher network bandwidths of the future will require.

A solution in which the two types of service (real-time and non-real-time) would be more intimately integrated is one based on packet switching, for example, of the fast packet switching variety mentioned above. But can such a network guarantee performance? This is the main question we plan to answer in this paper, by trying to provide a constructive proof of feasibility.

The first problem to be considered is whether a real-time service can be built at the transport or higher level in the protocol hierarchy on top of a datagram service. We believe that packet delay is much harder to control if each packet going from a given sender to a given receiver can in principle follow any route, and some of these routes may become congested while one or more real-time packets are in transit. In this paper, we therefore plan to restrict our attention to the case in which all packets belonging to a real-time channel always follow the same route. A possible approach, based on source routing, is one in which the source selects the path to each destination and keeps it fixed for the duration of the communication. However, the same result can be achieved by building real-time services on top of a connection-oriented (virtual-circuit) service offered by the network layer. The latter approach will more easily permit reservation of resources in each node along the route, which we believe is necessary to guarantee delay bounds. Even the *flow* abstraction in [Come88] uses a fixed route scheme with resources reserved in advance, in

spite of the authors' declared preference for a connectionless approach.

A major drawback of connections is that they have to be established before any communication can take place; this may, on the one hand, require a time longer than the client is able to wait in some real-time applications, and on the other hand make the shipment of a small amount of urgent information by a real-time service slower than by normal datagrams. In the study described here, we have tried to minimize the call setup time by designing an efficient establishment scheme.

3. A Formulation of the Problem

As discussed in the previous section, we study the feasibility of supporting performance guarantees in a wide-area network consisting of store-and-forward nodes interconnected by bounded-delay links. The network is assumed to offer connection-oriented service (as well as a connectionless service for those non-real-time communications where this service is preferable) at the network layer.

To formulate the problem in more detail, we will refer to the *parametrized message channel* (or simply *channel*) abstraction introduced in the design of the communication system of DASH [Ande88b]. DASH is a distributed operating system kernel being developed at the University of California at Berkeley for the very large distributed systems of the future, in which real-time communication requirements are expected to be important and widespread [Ande88a]. A channel (called *real-time message stream* or *RMS* in [Ande88b]) is a simplex connection between a sender and a receiver, which delivers messages (or packets) in sequence and is characterized by a number of parameters. The client (i.e., the entity that requests the establishment of a channel) specifies the values of the parameters to communicate its needs to the service provider, i.e., ultimately, to the network layer.

We are concerned here only with the performance-oriented parameters of a channel:

- the channel's capacity, defined as the maximum amount of information that may be outstanding at any given time in the channel;
- the maximum packet size;
- the delay bound or bounds for the channel's packets;
- the maximum packet loss rate.

While capacity and size are to be enforced by the client, delay bound and loss rate, whose values are to some extent interdependent [Sumi88], are to be guaranteed by the provider.

There are various types of channels, corresponding to the different types of delay bounds. In [Ande88b], the following bounds are considered:

- *deterministic*: the bound D is an absolute one; this is necessary in hard real-time applications;
- *statistical*: the bound is expressed in statistical terms; for instance, the probability that the delay of a packet is smaller than the given bound D must be greater than Z;
- *best-effort*: the bound D is not guaranteed by the provider, which, on the other hand, promises to do its best to satisfy it.

Note that the delay guarantees are assumed to be valid only for those packets that reach the destination. Delay bounds will not apply to packets that are allowed by the given maximum packet loss rate to go undelivered due to buffer overrun or to failures affecting any part of their channel. We also assume that the channels to be established are statistically independent of all the channels sharing totally or partially their route. If there are dependencies between a new channel and some of the existing ones, the client requesting the creation of the new channel should inform the provider. In this paper, we will not deal with the case in which there are dependencies

among channels.

The channel abstraction will be offered in the DASH network architecture by each layer to the layer above it, starting with the network layer. We are primarily concerned, in this paper, with the feasibility of establishing network-level channels that are "correct", i.e., that provide the required guarantees. Higher-level channels will certainly be implementable if we succeed.

Two important characteristics of the network are to be discussed before describing our approach: flow control and packet scheduling.

First of all, should flow control be window-based or rate-based? Both approaches to flow control seem feasible, but the rate-based one (in which the sender controls its packet sending rate on the basis of its knowledge of the characteristics of the receiver and of those of the channel's path) looks more attractive. Indeed, this solution does not require flow control acknowledgements, which would have to use another simplex channel, and would therefore be quite expensive. Alternatively, they could be sent as datagrams, in which case they may incur a rather large and perhaps highly variable delay. This delay might be too large, especially in long-distance transmissions, to be compatible with the frequencies and regularity of packet generation in most video or audio communication.

Rate-based flow control is feasible because, at channel establishment time, the appropriate resources can be committed, and, in particular, the receiver can check whether it will be able to accept packets at the rate declared by the sender. This verification is made possible by the deadline scheduling policy to be used in the receiver as well as in all the nodes (see below); in fact, the ability of the receiver to handle yet another stream of arriving packets can be verified with the same method (described in Section 4) used to test the ability of each intermediate node.

If flow control is rate-based, it is more convenient to replace the capacity parameter with one or more parameters describing the packet arrival (i.e., input) process. Here, we adopt for this purpose the parameters x_{min}, the minimum packet interarrival time on the channel, and x_{ave}, the minimum value of the average packet interarrival time over an interval of duration I. In other terms, x_{ave} is the average interarrival time during the channel's busiest interval of duration I. The ratio between these two parameters is the simplest possible measure of burstiness for the incoming packet stream. The amount of information "stored" in a channel is indeed hard to control on the part of the sender without a sliding-window mechanism, especially in a statistical channel, and even more in a best-effort one; the sender, on the other hand, has full control over, or at least full knowledge of, the packet generation rate, and can make sure that $1/x_{min}$ is never exceeded.

There is, however, the danger that a malicious user will circumvent any flow control mechanism and send into the network packets at a much higher rate than the declared maximum value, $1/x_{min}$, or maximum average value, $1/x_{ave}$. Indeed, current (and future) distributed systems include personal machines, whose operating system can easily be modified or even replaced by their owner or by an occasional user. The same effect might be caused by a failure in the sending host, even when this is a multi-user, protected-kernel system. If we do not take appropriate countermeasures, such malicious or faulty behavior can prevent the satisfaction of the delay bounds guaranteed to other clients of the real-time service, thereby damaging the clients and destroying the credibility of the service. Thus, a distributed flow control scheme seems absolutely necessary; with the characterization of the input we have selected, the scheme will have to make sure that neither $1/x_{min}$ nor $1/x_{ave}$ are exceeded. One possible solution to this problem is presented in Section 4.

Scheduling in the hosts and in the nodes will be deadline-based. More precisely, we adopt a modification of the EDD (Earliest Due Date) policy that gives, in the case of a conflict, priority to deterministic over statistical channels, and to statistical over best-effort channels. All of the other tasks of a host or node, including sending, forwarding, and receiving datagrams, have an even lower priority and are preemptable by real-time packets. Our scheduling policy is summarized in Figure 1.

4. An Approach to the Solution

The channel establishment mechanism we propose for the type of network described in Section 2 may be built on top of any procedure that can be used to set up connections. It is reasonable to design the routing algorithm for channel establishment so that it will exploit any available information about the delays along the various possible paths to the destination and about the "real-time load" of neighboring nodes. Besides trying to establish a connection, the mechanism will perform several tests and tentatively reserve resources in each node visited by the establishment request message. In order to make channel establishment fast, we impose on our procedure the restriction that it require only one round trip. Thus, the destination host is the last point along the path where the acceptance/rejection decision for a channel request can be made. When a node is revisited by an establishment message during this message's return trip, the resources previously reserved there must be either committed or released; hence a final, irreversible decision must have already been made.

The tests to be done in each node are concerned with the availability of sufficient bandwidth in the links, as well as processing power and buffer space in the node; that is, with determining whether the new channel can go through the node without jeopardizing the performance guarantees given to the already established channels passing through the same node.

If any test fails at a node, the channel cannot be established along that route; the message will be sent back, either to the sender (which may then decide to wait or try another output link) or to an intermediate node that can try sending the message towards the destination along another path. When an unsuccessful establishment request message revisits a node on its way back to the source, it frees all the resources that were tentatively reserved there during its forward trip.

If all the tests succeed at all nodes and at the destination host, this host subdivides the delay bound D (and the probability Z of not exceeding the bound if the channel being established is statistical) among the nodes traversed by the channel, after subtracting the total delays in the links along the route. Let d be the maximum delay bound assigned to the channel in a node, and z the minimum probability that d will not be exceeded in that node. A packet traveling on that channel and arriving at that node at time t will usually be assigned a node deadline equal to $t+d$. To satisfy the overall delay bound D, it is sufficient (though not necessary) to satisfy the bound d (or d, z) in each node along the route. For simplicity, we assume that satisfying this sufficient condition is the goal to be met by the establishment scheme in each node.

If all tests succeed, a reply message is sent back to the source host along the channel's route; this message notifies each node about the delay bound that has been assigned to it for the new channel, and commits all the reserved resources. The delay bound assigned to a node is then used to compute the deadline in that node for each packet traveling on that channel (see the calculation of *dl* in Figure 2). When the reply message reaches the source host, this host learns that the requested channel has been set up, and can start using it.

Note that, in the procedure just described, the destination host assigns delay bounds to nodes since we assume that the source host does not know how long the path to the destination will be. If the number of hops were known at the source beforehand, this assignment could be done by the source host, but it would be done less effectively, due to the source's incomplete knowledge of each node's loading situation.

Before discussing the tests to be performed in each node, we have to choose a suitable flow control mechanism. One possible approach, whose execution costs are to be carefully investigated, consists of increasing the deadlines of the "offending" packets, so that they will be able to go through lightly loaded nodes fairly rapidly, but will be delayed in heavily loaded nodes, where they would seriously interfere with the operation of other channels. When buffer space is limited, some of them might even be dropped because of buffer overflow. Of course, at least some of the buffers will have to be statically allocated to prevent the offending packets from flooding the

buffer space of a heavily loaded node and causing packets from other channels to be dropped.

One simple algorithm to increase the deadlines of packets arriving too soon after their predecessors is shown in Figure 2. The algorithm does not need any timers, which may be expensive to maintain. Note that the actual intervals between successive arrivals of a channel's packets at a node may be occasionally shorter than x_{min}, and the averages of the same intervals over I occasionally shorter than x_{ave}, either because of a sudden decrease of the load on the previous node along their path or (if the node in question is the first on the path) because of a higher packet rate illegitimately generated by the sending host. However, the distributed flow control algorithm in Figure 2 acts on the packets' deadlines so as to impose on the node the same switching and transmission load as if the packet stream on the channel obeyed the x_{min} and x_{ave} constraints. We can therefore assume that each channel going through the node satisfies those constraints.

To simplify our discussion, we shall assume here that the buffer space available for real-time connections in the network's hosts and nodes is unlimited, and that the network's error rate is always lower than the acceptable loss rates. This assumption will allow us to ignore the loss rate parameter, and to postpone the treatment of buffer space allocation and management to a subsequent paper.

When a node receives an establishment request message, it performs two or all three of the following tests:

(a) the *deterministic test*, required only when the channel to be established is deterministic, and involving only the deterministic channels already existing in the node;
(b) the *statistical test*, required for the establishment of both statistical and deterministic channels if at least one statistical channel is present in the node or is to be set up, and involving all deterministic and statistical channels in the node;
(c) the *delay bound test*, which is needed in all cases, and, if successful, is followed by the computation of the minimum delay bound for the channel in the node.

If the request passes the tests, then the node performs a *delay bound computation*, and then sends the establishment message on to the next node.

Note that best-effort channel requests are not subjected to any tests. Hence, they are always accepted. In the sequel, unless explicitly specified, by the term "channel" we shall refer only to deterministic or statistical channels.

The **deterministic test** consists of verifying that enough processing power is available in the node to accommodate the additional deterministic channel without impairing the guarantees given to the others. Since we are dealing with deterministic bounds, this must be true even in the worst possible case, i.e., even when all deterministic channels are sending packets into the node at their maximum rates. The maximum utilization of a node by channel i, whose packets have a maximum service time in the node equal to t_i, is $t_i / x_{min,i}$. Thus, the condition to be tested is:

$$\sum_j t_j / x_{min,j} \leq 1, \qquad (1)$$

where the sum extends to all deterministic channels, including the one to be established (channel i).

The **statistical test** has two purposes:

(i) to determine whether for each statistical channel j in the node the probability of a delay higher than the bound d_j is below its maximum tolerable value, $1 - z_j$;
(ii) to provide the destination host with the information that is necessary to compute z_i for the new channel if this is statistical.

Both these purposes can be obtained by computing the *probability of deadline overflow* P_{do}. A packet may be delayed beyond its delay bound in a node because of two reasons: a temporary saturation of the node's processing and switching capacity (*node saturation*), and impossible scheduling constraints (*scheduler saturation*). The latter is avoided, as discussed below (in our description of the delay bound test), whereas the statistical test deals with the former.

The probability that channel j is active during an interval I is

$$p_j = x_{min,j}/x_{ave\,j}. \tag{2}$$

Given m independent channels 1, 2, ... m passing through a node, the probability that k of them (say, the first k) are simultaneously active is given by

$$Prob(1,2,..k,\neg k+1,..\neg n) = \prod_{j=1}^{k} p_j \prod_{l=k+1}^{m} (1-p_l). \tag{3}$$

Let at least one of the channels be statistical. To compute P_{do}, we start by listing all the *overflow combinations*. An overflow combination is a set of channels that, when simultaneously active for a sufficiently long time, will cause packets to miss their deadlines. In other words, an overflow combination is one for which inequality (1) above, with the sum extended to all active channels of both types, is not satisfied.

Let H be the set of overflow combinations in the node, h be a member of H, and $P(h)$ be the probability of occurrence of combination h computed as in (3) above. Then,

$$P_{do} = \sum_{h} P(h). \tag{4}$$

We can then check whether the following inequalities are satisfied:

$$P_{do} \leq 1 - z_j, \tag{5}$$

for all statistical channels j existing in the node. To see whether (5) is satisfied, it is sufficient to verify that

$$P_{do} \leq \min(1-z_j), \text{ or } 1-P_{do} \geq \max z_j. \tag{6}$$

If test (6) is successful, then the value of P_{do} is sent to the destination host for the purpose referred to in (ii) above. The destination host will try to assign to channel i in the node a value of z_i that satisfies inequality (6).

Figure 3 shows two examples of statistical test. The node being considered has four channels (numbered 1 through 4) already established when it receives a message requesting the establishment of channel 5. This is a deterministic channel whose addition would cause node saturation whenever all channels have maximum rate traffic on them (see Figure 3(a)). Since, as shown in Figure 3(b), the probability of deadline overflow is 0.000161, and its complement to 1 (0.999839) is greater than the z's of the two statistical channels (channels 2 and 3), the node passes the statistical test. The value of P_{do} is not transmitted to the destination host in this case, as channel 5 is deterministic.

The same figure also shows in (a) the characteristics of another new channel, called channel 6, and in (b) the calculation of P_{do} according to equations (3) and (4), under the assumption that channel 5 has been established. The node passes the statistical test also for channel 6, and transmits the value of P_{do} (0.000943) to the destination host, since channel 6 is statistical.

The **delay bound test** determines whether scheduler saturation can be avoided in the node, and, if so, the minimum delay bound to be assigned there to the channel being established so that this goal will be achieved. We try to eliminate this type of saturation primarily because of the complexity of dealing with it in the general case. A simple example of scheduler saturation is that of two channels with respective node service times 3 and 4 units, and respective node delay bounds 5 and 6 units; if two packets from the two channels arrive simultaneously at the node, the scheduler will in no way be able to schedule them so that their deadlines are met.

To determine whether scheduler saturation is possible in a node, we divide the m (deterministic or statistical) channels passing through the node into two sets: we call A the set of those channels whose delay bound in the node is lower than the sum of all channel service times, and B the set of those channels whose delay bound in the node is greater than or equal to that sum:

$$A=\{i \mid i=1,...a;\ d_i < \sum_{j=1}^{m} t_j\}, \tag{7}$$

$$B=\{l \mid l=a+1,...m;\ d_l \geq \sum_{j=1}^{m} t_j\}, \tag{8}$$

With no loss of generality, we number the a channels in A according to the order in which they would be scheduled by the algorithm in Figure 1 if they arrived all at the same instant: 1 will be the channel that would be shipped first, a the one that would be shipped last.

Let us assume that the node is not saturated: we have already taken into account node saturation, which delays packets beyond their bounds whether or not scheduler saturation is present, in the statistical test; also, let us assume that

$$x_{min,l} + d_l \geq \sum_{j=1}^{m} t_j + t_l \tag{9}$$

for $l=1,...m$. This assumption is equivalent to postulating that no subsequent arrivals of packets at the node will interfere with the transmission of those (one per active channel) we are considering. A discussion of how to proceed when the assumption is not satisfied can be found in the Appendix. The following theorem, on which we will base the delay bound test, holds.

Theorem 1: Scheduler saturation is impossible if and only if

$$d_i \geq \sum_{j=1}^{i} t_j + \max_{i<k\leq m} t_k, \quad (i=1,...a). \tag{10}$$

Proof: Sketched in the Appendix.

The following corollaries can be easily derived from Theorem 1:

Corollary 1: If B is empty ($a=m$) and A is non-empty ($a > 0$), there is at least one packet arrival pattern that causes scheduler saturation.

Proof: $d_a \geq \sum_{j=1}^{m} t_j$ contradicts (7).

Corollary 2: If B has only one member ($a=m-1$) and A is non-empty ($a > 0$), there is at least one packet arrival pattern that causes scheduler saturation.

Proof: $d_a \geq \sum_{j=1}^{m-1} t_j + t_m = \sum_{j=1}^{m} t_j$ contradicts (7).

Corollary 3: If A is empty ($a=0$), scheduler saturation is impossible.

Proof: Inequalities (10) are trivially satisfied in all cases for $a<i\leq m$.

The table in Figure 4(a) lists the four channels 1, 2, 3, and 4 in Figure 3(a), ordered by increasing delay bound. The fourth column (r_i) displays the value of the right-hand side of condition (10) for those channels that are members of A. Since $d_i \geq r_i$ for all i, there is no scheduler saturation. The addition of channel 5 causes node saturation whenever packets from all channels are simultaneously present in the node. To avoid scheduler saturation in all cases in which there is no node saturation, we have to apply the delay bound test to the worst non-saturated-node case. The worst case with respect to scheduler saturation is obtained by deleting channel 2 (the one with the smallest t) from the table in Figure 4(a). Figure 4(b) shows that channel 5 cannot be assigned a delay bound smaller than 11 time units, otherwise by Corollary 2 there would be scheduler saturation. Thus, the value of $min\ d_5$ to be transmitted to the destination host is 11.

The arrival of the request for the establishment of channel 6 forces us to drop another channel from the table in Figure 4(b), since 1, 3, 4, 5, 6 is one of the overflow combinations (see Figure 3(b)): assuming that channel 5 has been assigned a delay bound of 16 units in the node, we drop channel 1, so that channel 6 will be forced to join set B, and to have a larger value of $min\ d$. The resulting table is presented in Figure 4(c). Channel 6 can be assigned a minimum delay bound of 10 units, equal to the threshold that divides sets A and B in this case.

Figure 5 illustrates the establishment of channel 6 on a 3-node connection. The characteristics of node 1 (with channels 1 through 5, and $d_5 = 16$) are displayed in Figure 3(a); those of nodes 2 and 3 in Figure 5(a) and (b), respectively. Figures 5(c) and 5(d) show the tables built by the delay bound computations performed in node 2 and 3, respectively. Note that channel 6 saturates node 3, and therefore channel 10 is eliminated from the table in Figure 5(d).

The destination host, if and when it receives the establishment request message, is to determine whether the total value of Z can be factored into node contributions z_k ($k = 1, 2, ...n$):

$$Z = \prod_k z_k, \tag{11}$$

The destination host can answer the question about whether Z can be factored as in (11) by checking whether

$$\prod_k (1 - P_{do,k}) \geq Z. \tag{12}$$

If (12) does not hold, then the request is rejected. Note that (11) implies the assumption that, once a packet is delayed beyond its deadline in a node, it will not be able to satisfy the channel's overall delay bound. This assumption is justified by the behavior of a saturated node: even if the delay bounds are long, but finite, packets will eventually be delayed beyond them, and the delays will grow without bounds while the node is saturated. If, however, the saturation condition is short, some or all of the deadlines might still be met. This, and the assumption also implied by (11) that each node will delay, if any, only previously undelayed packets, are certainly pessimistic assumptions. Rejecting a request if (12) does not hold is a policy based on worst-case considerations, which guarantee that the desired results will be achieved in all possible circumstances. Of course, there is a cost associated with worst-case design: these and other conservative decisions reduce the maximum number of channels that can be established in a given network with respect to that which the same network could support if the design of the real-time service were less conservative. The two important questions this observation raises are (i) how large this reduction is going to be in practice, and (ii) whether a less conservative approach could offer similar performance guarantees. Question (i) will be answered by the ongoing simulation effort; question (ii) will be the subject of a future investigation.

In the example illustrated in Figures 3(a), 4(c), 4(d), and 5, the destination host receives the information summarized in Figure 6. Two very simple algorithms to compute the bounds for channel 6 packets in each node and their results are shown in the same figure. Channel 6's delay bound requirements were $D = 30$ and $Z = 0.98$. (Note that D is assumed to be completely assignable; that is, such fixed and known delays as the propagation time are not included in the value of D; this means that the actual delay bound specified by the client for channel 6 is higher than D, unless propagation delays are negligible.) Since it is possible to satisfy the requirements, the channel establishment request is accepted.

This concludes the description of our establishment algorithm. The algorithm can be improved in several ways: for example, a method, even an approximate one, which made the calculation of P_{do} faster while retaining the correctness of the whole approach would make channel establishment faster even in those cases where performing the statistical test requires the enumeration of overflow combinations.

5. Conclusion

The goal of this study was to determine the feasibility of offering real-time services in packet-switching wide-area networks. We have defined the problem and the type of network to be studied, adopted the parametrized message channel abstraction, selected several types of

performance guarantees such a service could provide, specified the meaning and extent of these guarantees, and selected a simple characterization of a channel's input packet stream. We have argued for connection-oriented packet switching as the basis of a real-time service, as it seems very difficult or impossible to build channels on top of a connectionless service. We have also claimed that connections (and performance guarantees) must be provided at the lowest level in the protocol hierarchy (i.e., at the network layer) in order to make those guarantees available at the higher levels.

A single-round-trip procedure for establishing channels has been devised. The procedure entails several tests and tentative reservations of resources to be performed in each node along the channel's path. A channel that passes these tests can be guaranteed the type and value of delay bound requested by the client; these guarantees become effective as soon as the channel is established and remain in effect until it is disconnected. Performance can be guaranteed because of the scheduling and distributed flow control policies adopted as well as because of the worst-case bounding arguments our establishment scheme is based on.

Many problems remain to be explored in the area of wide-area real-time communication:

- buffer allocation and management policies suitable for the type of real-time service described in this paper;
- the best ways to guarantee a given bound on delay variance or jitter;
- the possibility of devising correct algorithms for the establishment of channels whose traffic is described by parameters different from x_{min} and x_{ave}, or is not independent of the traffic on other channels;
- the introduction of security, fault tolerance, accounting, and charging capabilities into the design of a real-time service;
- a procedure to be used for *fast channel establishment*, i.e., for setting up a channel while delivering the first packet on that (not yet existing) channel to the destination;
- the alternative ways of providing performance guarantees in a wide-area network, and their advantages and disadvantages with respect to the approach discussed in this paper;
- the feasibility of implementing real-time services on an Ethernet, or in a virtual-circuit network consisting of Datakit† [Fras83] or Datakit-like nodes;
- the implementability in hardware of the node functions required for real-time services.

Acknowledgements

A large number of individuals have contributed to the ideas introduced in this paper and to their presentation. David Anderson proposed the channel abstraction, which is the basis of the approach to real-time communication described above. Many of the issues and solutions discussed in the paper resulted from conversations with Dinesh Verma and Kang Shin. Dinesh Verma revealed by his simulation experiments a number of interesting and unexpected phenomena that had to be taken into account in the design of the scheme. Parts of the manuscript were read by Ramon Caceres, Caryl Carr, Sandy Fraser, Riccardo Gusella, Sam Morgan, Dave Presotto, Keshav Srinivasan, Kang Shin, and Dinesh Verma, who provided very helpful comments. Peter Danzig's many constructive suggestions improved very substantially the presentation. The members of the DASH Project contributed feedback and support throughout the effort. The errors and omissions that, in spite of all the help he received, can still be found in the paper are an exclusive intellectual property of the author.

† Datakit is a trademark of AT&T Bell Laboratories.

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Appendix

Theorem 1: In a non-saturated node with

$$x_{min,l} + d_l \geq \sum_{j=1}^{m} t_j + t_l, \ (l=1,...m), \tag{A.1}$$

scheduler saturation is impossible if and only if

$$d_i \geq \sum_{j=1}^{i} t_j + \max_{i<k\leq m} t_k, \ (i=1,...a). \tag{A.2}$$

Proof: Since we have excluded the possibility of node saturation even in the worst case (i.e., when all channels are carrying packets at their maximum rates $1/x_{min,j}$), there cannot be any buildup of queues in time; we can therefore assume that the node is empty when we start examining arrival patterns, and call time 0 the instant at which the first packet arrives at the node. Arrival times can be assumed to be arbitrary, since channels are supposed to be independent of each other; the dependencies that may result from the sharing of an input link by two or more channels can only improve the situation, as this sharing serializes arrivals on those channels at the node. Packets arriving on a given channel after the first we consider are not independent of that first: the second packet on channel j will in the worst case arrive $x_{min,j}$ time units after the first packet on the same channel. Because of assumptions (A.1), no deadline for a subsequent packet will fall within the interval between time 0 and time $\sum t = \sum_{j=1}^{m} t_j$. Subsequent packets can therefore be ignored in this proof.

(i) *Only if* part. If condition (A.2) is not satisfied for some i, there is at least one arrival pattern that causes scheduler saturation. Let a packet with service time $\max_{i<k\leq m} t_k$ arrive at time 0, and packets from all other channels in A arrive at time 0+. Then, the packet on channel i will be completely shipped only at time $\sum t + \max_{i<k\leq m} t_k$, which is greater than its deadline d_i.

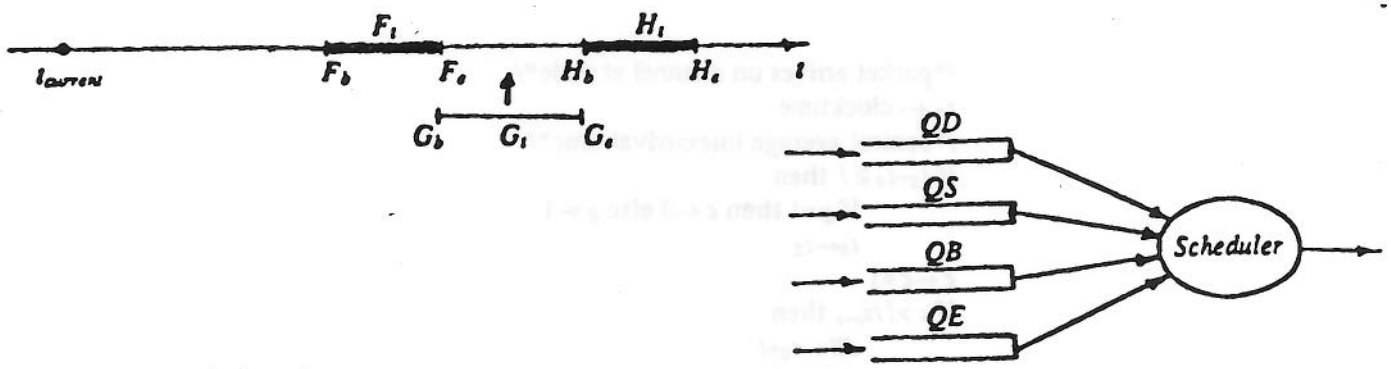
(ii) *If* part. If conditions (A.2) are all satisfied, there cannot be scheduler saturation. Let the packet on channel i arrive at time 0; in this case, its latest possible departure time is $\sum_{j=1}^{i} t_j < d_i$. If it arrives at 0+, the latest departure time is $\sum_{j=1}^{i} t_j + \max_{i<k\leq m} t_k \leq d_i$. With an arrival time between 0+ and $\max_{i<k\leq m} t_k$, the maximum time spent by the packet in the node is less than d_i. Arriving at $(\max_{i<k\leq m} t_k)+$, this maximum time is $\sum_{j=1}^{i} t_j < d_i$, and later arrivals yield even smaller maximum times.

Q.E.D.

If condition (A.1) is not satisfied for all channels, we apply Theorem 1 to a set of channels that includes some subsequent packets. Since considering these packets increases $\sum t$, we calculate the new value of this sum, $(\sum t)'$, from the following recursive definition:

$$(\sum t)' = \sum t + k_h t_h,$$
$$k_h = 1 \text{ if } x_{min,h} + d_h < (\sum t)', \tag{A.3}$$
$$k_h = 0 \text{ otherwise.}$$

Sets A and B are defined by the value of $(\sum t)'$. The arrival times of some of the packets to be considered are not independent of those of the others. However, it is easy to see that the independence assumption is, once again, worst-case (as it may declare scheduler saturation to be possible when instead it is not). Applying Theorem 1 to this case while ignoring dependencies is therefore safe, since it does not endanger any of the performance guarantees provided to the clients of the service.



at arrival of packet A:
select appropriate Q
insert (A,Q)
return

upon completing the processing of a packet:
D | S | B | E ← *head* (QD | QS | QB | QE)
if D | S | B = *nil* then
    D_b | S_b | B_b ← ∞
    D_e | S_e | B_e ← ∞
if D_b | S_b | B_b < $t_{current}$ then
    D_b | S_b | B_b ← $t_{current}$,
    D_e | S_e | B_e ← (D_b+D_t) | (S_b+S_t) | (B_b+B_t)
if D_b < S_e then
    send D
    return
else if S_b < B_e then
    send S
    return
else if B ≠ *nil* then
    send B
    return
else if E ≠ *nil* then start E until arrival
  of next real-time packet
return

insert (G,Q)
find correct place of G in Q
F ← left packet
H ← right packet
place G between F and H
if Q ≠ QD then return
align (G,H)
align (F,G)
if F_m = 1 then
    F_m ← 0
    insert (F,QD)
return

align (B,C)
if $C_b \le B_e \le C_e$ then
    $B_e \leftarrow C_b$
    $B_b \leftarrow B_e - B_t$
    $B_m \leftarrow 1$
else if $C_b \le B_b \le C_e$ then
    $C_e \leftarrow B_b$
    $C_b \leftarrow C_e - C_t$
    $C_m \leftarrow 1$
return

Figure 1. Scheduling policy for hosts and nodes. The four queues are for deterministic (QD), statistical (QS), and best-effort (QB) packets, and for everything else (QE). The notation statement (D | S | B | E) is a shorthand for: statement (D); statement (S); statement (B); statement (E). F, G, H are deterministic packets; their end times F_e, G_e, H_e coincide with those called *dl* in Figure 2, and their node service times are denoted by F_t, G_t, H_t. Boolean variable F_m, if 1, indicates that F has been moved leftward on the time axis.

```
/*packet arrives on channel at node*/
t2 ← clocktime
/*control average interarrival time*/
if t2−t0 ≥ I then
        if g=1 then c←0 else g←1
        t0←t2
c←c+1
if c >I /xave then
        dll←t0+I
        dd←dll−t2+d
        c←0
        g←0
/*control minimum interarrival time*/
if t2−t1<xmin then t1←t1+xmin else t1←t2
/*set packet deadline*/
dl←t1+d
if dl≤dll then dl←t1+dd
dll←dl
return
```

---

t_0= starting time of channel's current I interval
t_1= arrival time of previous packet on channel
c = packet counter for channel (initially 0)
d = maximum delay for channel's packets in node
dl = packet deadline (time it must have left node)
x_{min}, x_{ave}, I = input characteristics of channel
g = termination cause indicator (0: excess packets in I;
1: I interval expiration); initially: g = 1.
dll = auxiliary variable; initially: dll=0.
dd = extended deadline when excess packets in I; initially: dd=0.

Figure 2. Distributed flow control algorithm. It is executed before the *at arrival of packet A* algorithm in Figure 1.

Channel #	Type	x_{min}	x_{ave}	t	d	z	t/x_{min}	p
1	D	10	35	3	14	-	0.3	0.2857
2	S	4	30	1	6	0.99	0.25	0.1333
3	S	15	90	2	8	0.982	0.1333	0.1667
4	D	8	70	2	8	-	0.25	0.1143
5	D	20	90	4	?	-	0.2	0.2222
6	S	8	70	2	?	?	0.25	0.1143

(a)

1	2	3	4	5	6	$\sum$	Prob	P_{do}
+	+	+	+	+		1.1333	0.000161	0.000161
+	+	+	+	+	+	1.3833	0.000018	
∅	+	+	+	+	+	1.0833	0.000045	
+	∅	+	+	+	+	1.1333	0.000119	
+	+	∅	+	+	+	1.25	0.000091	
+	+	+	∅	+	+	1.1333	0.000142	
+	+	+	+	∅	+	1.1833	0.0064	
+	+	+	+	+	∅	1.1333	0.000142	
+	+	∅	+	∅	+	1.05	0.000322	0.000943

(b)

Figure 3. Examples of statistical test in a node with channels 1, 2, 3, 4 for the establishment of channel 5, and in the same node with channels 1, 2, 3, 4, 5 for the establishment of channel 6: (a) channel characteristics; (b) overflow combinations and calculation of P_{do}.

Channel #	t_i	d_i	r_i
2	1	6	4
----	----	----	----
3	2	8	
4	2	8	
1	3	14	

(a)

Channel #	t_i	d_i	r_i
4	2	8	6
3	2	8	8
----	----	----	----
1	3	14	
----	----	----	----
5	4	?	

(b)

Channel #	t_i	d_i	r_i
4	2	8	6
3	2	8	8
----	----	----	----
5	4	16	
----	----	----	----
6	2	?	

(c)

Figure 4. Tables for the delay bound test and the delay bound computation: (a) all the original channels (1 through 4 in Figure 3(a)); (b) addition of channel 5 (and deletion of channel 2); (c) addition of channel 6 (and deletion of channel 1). Note that the channel # in the tables is the same as in Figure 3(a); the channels in set A in each table are ordered as explained in the description of the delay bound test.

Channel #	*Type*	x_{min}	x_{ave}	t	d	z	t/x_{min}	p
7	*D*	*15*	*50*	*4*	*12*	*-*	*0.2667*	*0.3*
8	*S*	*18*	*80*	*6*	*15*	*0.993*	*0.3333*	*0.22*

(a)

9	*D*	*15*	*50*	*4*	*16*	*-*	*0.2667*	*0.3*
10	*D*	*10*	*70*	*2*	*14*	*-*	*0.2*	*0.1429*
11	*S*	*10*	*60*	*3*	*20*	*0.985*	*0.3*	*0.1667*

(b)

Channel #	t_i	d_i	r_i
7	4	12	
8	6	15	
----	----	----	----
6	2	?	8

(c)

Channel #	t_i	d_i	r_i
9	4	16	
11	3	20	
----	----	----	----
6	2	?	6

(d)

Figure 5. Characteristics and delay bound tests for the two additional nodes traversed by channel 6 (see Figure 3(a)) to reach the destination host: (a) node 2; (b) node 3; (c) table for node 2; (d) table for node 3.

Node	min d	max z	d	z
1	10	0.99905	12	0.9929
2	8	1	10	0.9939
3	6	0.99918	8	0.9931

$$D - \sum(\min d) = 6;\ d_i = (D - \sum(\min d))/n + (\min d)_i$$

$$g = Z/\prod(\max z) = 0.9817;\ z_i = g^* \exp(1/n)(\max z)_i$$

Figure 6. Information and algorithms for the destination host. The results produced by the algorithms are displayed in the d and z columns. n is the number of nodes traversed by the channel being established (in the example, $n = 3$).