Fast Low-Cost Failure Recovery for Reliable Real-Time Multimedia Communication

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Abstract

Transmission of multimedia data over a packet-switched network typically requires resource reservation to guarantee an acceptable level of performance (e.g., throughput or delay). In this article we address the problem of how to make such real-time communication reliable. First of all, it is essential to bound the duration of service disruption caused by failures to a reasonably small value. Considering the large volume of multimedia data, minimizing the fault-tolerance overhead is also important. Furthermore, as more applications with different dependability requirements share the same network, the level of dependability for a given application should be "customizable," depending on the criticality of the application. We first survey the existing approaches, and then present our scheme which is developed in accordance with three design goals: fast failure recovery, low fault-tolerance overhead, and per-connection reliability guarantee. Our scheme provides an integrated solution covering such issues as connection establishment, failure detection, runtime failure recovery, and resource reconfiguration.

Real-time transport of continuous media (i.e., video and audio) has traditionally been achieved by circuit switching in telephone services or by broadcasting over shared media in television services. However, it is difficult to realize real-time continuous multimedia applications on packet-switched computer networks, since the end-to-end packet delay and throughput of a media stream are inherently nondeterministic. Such end-to-end performance, necessary to achieve the required functionality of these applications, is often called end-to-end quality of service (QoS). Today's representative computer network, Internet, also lacks QoS support for continuous media applications; the window-based flow control is unsuitable for traffic with end-to-end timing constraints. Nevertheless, several multimedia applications have already been deployed on Internet using such protocols as RTP [1], XTP [2], and IP multicast. However, these protocols do not meet the true multimedia requirements because they only support a best-effort service model. The next-generation Internet is expected to provide new services that meet the diverse QoS requirements of various emerging multimedia applications.

In response to this growing demand for real-time communication services, considerable efforts have been made in recent years and numerous QoS service models developed, ranging from static constant bit rate (CBR) service, which resembles the telephony service, to the "controlled-load service," which mimics the best-effort service in unloaded networks [3, 4]. Unlike traditional datagram services in which average performance is of prime interest, guaranteeing QoS is the key requirement of the real-time communication service. To achieve QoS guarantees, most real-time communication schemes rely on some form of resource reservation and admission control, while each differs in QoS parameters and/or firmness of QoS guarantees. That is, they share three common properties: they are *QoS-contracted*, *connection-oriented*, and *reservation-based*. A contract between a client and the network is established before the client's messages are actually transferred. To this end, the client must first specify his input traffic behavior and required QoS. The network then computes the resource needs (e.g., link and CPU bandwidths, and buffer space) from this information, selects a path, and reserves necessary resources along the path. If there are not enough resources to meet the client's QoS requirement, the request is rejected. The client's data messages are transported only via the selected path with the resources reserved, and this virtual circuit is often called a *real-time channel.*

Network Dependability

Primitive real-time communication services will soon be available for such multimedia applications as Internet phone, the Web, and digital libraries. For example, the controlled-

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load service is considered a possible candidate. On the other hand, the increase of network connectivity and link capacity will expand the application domain of real-time multimedia communication to include business- or mission-critical applications, such as remote medical service, collaborative scientific research, business net-meeting, real-time electronic commerce, or even remote battlefield command/control. Such critical applications require *both* dependable and timely communication services. Suppose, for example, there is a very important videoconference, and unanticipated network failures disconnect one or more participants from the conference for an unpredictably long period. This may lead to a failure or delay in reaching important strategic decisions, which can cause significant economic loss. In this example the connection availability, defined as the probability of a connection being available at any given time, is a key dependability QoS measure.

Network failures can cause an even larger-scale social disaster. Catastrophic social consequences of network failures have actually been witnessed in recent breakdowns of the U.S. telecommunication network. For instance, a fire at an unmanned tall office building in Illinois caused 3.5 million telephone calls to be blocked in 1988. Emergency 911 calls went unanswered, online business transactions were stopped, and flights were delayed because of air traffic control failures. Hospital operations were affected, and drug stores could not process prescriptions. Even banks had to be closed for security reasons due to disabled alarm systems [5]. In the 1990s several similar accidents have been reported for various reasons, such as damage of a fiber cable caused by construction, earthquakes, outage of switching systems, or network overload. Although network failures rarely occur, the consequence of mishandling failures could be devastating, thus making network reliability a major concern.

The current Internet with datagram services has successfully dealt with two types of network failures: *transient* and *persistent*. A typical example of a transient failure is temporary packet losses due to either network congestion or data corruption. Persistent failures include the breakdown or crashing of network components. Transport protocols such as TCP can handle transient loss of packets by acknowledgment and retransmission, and the connectionless IP protocol deals with persistent failures by routing packets around faulty network components. However, retransmission is unlikely to be useful for real-time communication, because there is usually not enough time to detect and retransmit a lost real-time message before its deadline expires. Instead, forward error correction (FEC) techniques should be used if no data loss is acceptable. The main drawback of FEC is its high overhead. We also face a serious difficulty in tolerating persistent failures for real-time communication, because a QoS guarantee is realized by reserving resources on a fixed path and transporting real-time messages only via the path. Hence, a real-time message, unlike a datagram message, cannot be detoured around faulty components on the fly.

The prevalence of optical fibers affects network dependability in two ways. First, the probability of transmission errors in optical links becomes negligible; the error rate is dropped from 10^{-5}/b in the 56 kb/s links of initial ARPANET to below 10^{-10}/b in optical links [6]. The chance of packet loss due to transmission errors is very rare, and most packet losses are attributed to congestion-control problems. As a result, for real-time communication tolerating transient failures has become relatively less important because congestion-induced packet losses can be avoided by resource reservation. Furthermore, occasional loss of messages is tolerable in many multimedia applications. In contrast, the deployment of optical fibers exacerbates the difficulty in tolerating persistent failures, because more connections will be running through each large-capacity link, and thus, even a single link failure can result in the loss of a large number of connections. Unless the network is carefully designed to restore and then deliver the large amount of traffic lost due to failures, the increase of link capacity will seriously threaten network dependability. Not only link failures but also node failures are getting more difficult to deal with. Usually, Internet routers are not designed to meet as stringent a reliability goal as telephone network switches (e.g., AT&T 5ESS). Higher-performance routers in the future Internet will become even harder to provide high reliability for, due mainly to their complex software. Moreover, computer networks are more vulnerable to vandalism such as viruses or hacking than the telephone network, which has a "closed" architecture.

Considering the criticality of network dependability and the increasing threat of network failures, the development of effective mechanisms to cope with network failures is a must.

Desirable Features

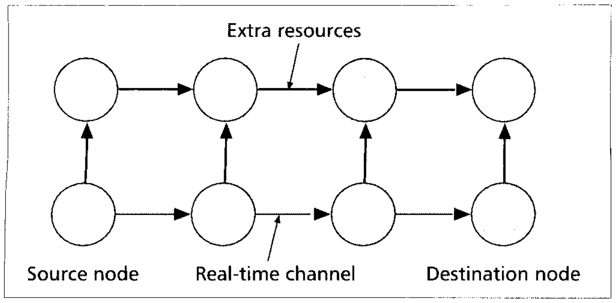
To design a fault-tolerant service, one must first define the model of failures to be tolerated. Some applications can tolerate slow failure recovery but require reliable (correct) delivery of all messages, even if it takes a long time. Examples of such applications are electronic mail and file transfer. For these applications, message-level failure handling is necessary. Thus, the receiver informs the sender of the reception of each message (or group of messages), so that the sender can detect delivery failures and retransmit lost messages. Some other applications require fast failure recovery, but data loss during failure recovery can be tolerated. Real-time multimedia applications fall into this category, since they do not require such strict reliability as "no message loss at all." For example, loss of a couple of frames in video/voice data streams may be acceptable. Therefore, the fault-tolerant service for multimedia applications should be designed with a different reliability goal than that of conventional non-real-time applications.

In this article we assume that (infrequent) transient packet losses are acceptable to applications, or are dealt with by other techniques like FEC, and focus on how to effectively handle "persistent" or "permanent" failures of network components (e.g., *crash failures*). Instead of directly modeling component failures, we define and use a new failure semantic, *channel failure*. A real-time channel is said to have "failed" if the rate of correct[1] message delivery within a certain time interval is below a threshold specified by the application; that is, the granularity of failures is "persistent disruption of channels" instead of "individual message loss." The same error rate may be acceptable to some applications, and not to others.

There are five criteria that characterize a good solution to this problem:

- Per-connection dependability guarantee — Each connection may request a different level of fault tolerance depending on its criticality. The network should provide guarantees on dependability for each connection, so that successful recovery is guaranteed as long as failure occurrences do not exceed the fault-tolerance capability of the connection.
- Fast (time-bounded) failure recovery — The service disruption time of a connection caused by failures should be bounded to a reasonably small value.

[1] *In terms of both content and timing.*



■ Figure 1. *An example of an SFI channel.*

- Small fault-tolerance overhead — The additional resource overhead required for fault tolerance should be acceptably low.
- Robust failure handling — Failures should always be handled robustly, even though failure occurrences may exceed the assumed failure hypothesis. By "robustly" we mean that the QoS of nonfaulty real-time channels is not affected, and as many faulty real-time channels as possible are recovered.
- Interoperability/scalability — The failure recovery scheme must be interoperable with various existing and future real-time channel protocols. Also, it should scale well in a dynamic environment where (short-lived) connections are set up and torn down frequently, and the distributions of connection setup requests and connection holding time are unknown.

Existing Approaches

Before presenting our approach, we will first survey some notable previous work on dependable real-time communication. Particularly, the schemes developed for computer and telephone networks are reviewed with comparisons against our approach.

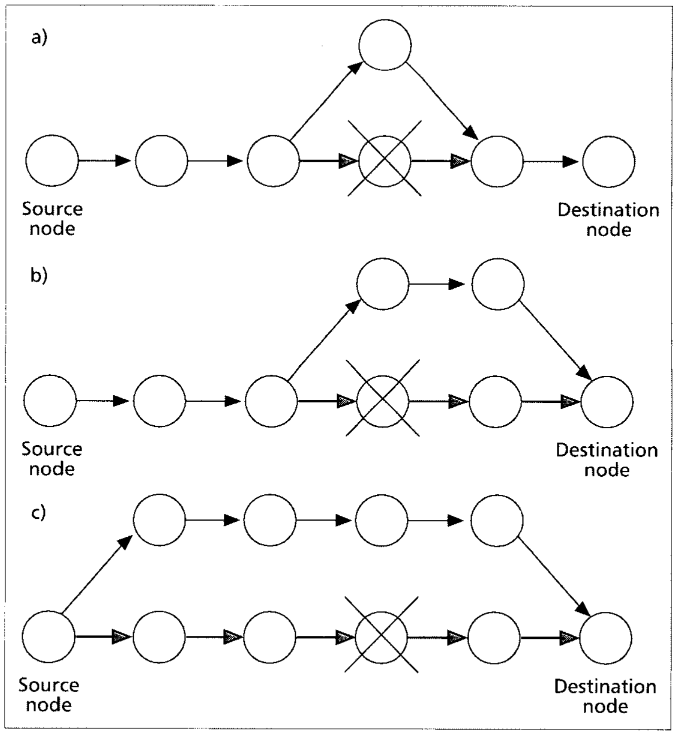
Packet-Switched Computer Networks

There have been roughly three types of approaches to the problem of achieving dependable real-time communication in packet-switched computer networks.

First, the simplest way of recovering a real-time channel from a network component failure is to establish a new real-time channel which excludes the failed component. This *reactive* method is studied in [7]. This scheme relies on the broadcast of all component failures to the entire network so that all hosts can maintain a consistent view of the current network topology. When a source node recognizes the failure of its channel from this broadcast, it tries to establish a new channel to replace the disabled one. Since no provisions are made a priori for fault tolerance, this method causes no fault-tolerance overhead in the absence of failures. However, it does not give any guarantee of failure recovery. The channel re-establishment attempt can fail due to resource shortage at that particular time. Even when there are sufficient resources, the contention among simultaneous recovery attempts for different faulty connections may require several trials to succeed, thus delaying service resumption and increasing network traffic. To regulate simultaneous recovery attempts, random delays can be introduced before starting each recovery operation.

The second approach is failure masking, in which multiple copies of a message are sent simultaneously via disjoint paths [8]. This method attempts to achieve both timely and reliable delivery at the same time. Thus, by transmitting multiple copies over disjoint paths for a message, the chance that at least one copy is delivered within its deadline increases, and the effects of possible failures are masked. This approach has an advantage in that both persistent and transient failures are handled without service disruption, but it is very expensive due to the additional resource consumption for transmitting multiple copies of the same message. To enhance resource efficiency, [9] presents a scheme which combines error coding with multiple-copy transmission. In this method, instead of transmitting multiple copies of an entire message, each message is broken into equal-size submessages which are then transmitted over disjoint paths. In addition, some redundant information is transmitted over separate paths for FEC. When some submessages are lost due to failures, the original message can be reconstructed using the redundant information. This method allows a trade-off between resource overhead and fault-tolerance capability by selecting the number of redundant information channels. However, this method still involves additional resource consumption for FEC, which is not necessary if the underlying multimedia applications can tolerate infrequent transient data loss. Another practical problem is that the connectivity of current (or near future) network topologies may not provide a sufficient number of disjoint paths between any two network nodes, which may hinder the flexibility of this method.

The third approach lies between the above two approaches in terms of fault-tolerance overhead. In this approach, cold-standby resources are reserved for fault tolerance. The Single Failure Immune (SFI) scheme [10] took this approach to provide guaranteed failure recovery under a single failure model. In this scheme, additional resources are reserved in the vicinity of each real-time channel at the time of channel establishment, and when a failure occurs the failed component is detoured by altering the channel path using the



■ Figure 2. *Three rerouting strategies: a) local rerouting; b) local-to-end rerouting; c) end-to-end rerouting.*

Recovery method	Resource overhead	Recovery delay	Recovery guarantee
Reactive	No	Long	No
SFI	High	Shorter	Deterministic
Multicopy	Very high	No	Flexible
Span restoration	Low	Shorter	Deterministic
Path restoration	Lower	Short	Deterministic
Our approach	Lower	Short	Flexible

■ Table 1. *A comparison of existing approaches.*

reserved resources. Figure 1 illustrates the setup of an SFI channel. The advantage of this cold-standby approach is that although additional resources need to be reserved, the resources reserved for fault tolerance can be utilized by best-effort traffic in the absence of failures. Another cold-standby method is presented in [11]. In this method, cold-standby reservation is combined with multiple-channel transmission. It differs from the multiple-channel FEC method in that extra subchannels remain as cold-standby in the absence of failures; thus, FEC is not provided. The cold-standby extra channels are activated to replace the original submessage channels disabled by failures.

Telephone Networks

In old telephone networks, two telephones were connected by a true electric circuit through electro-mechanical or pure electric exchanges. Even after the introduction of digital transmission hierarchies like synchronous digital hierarchy) (SDH) and synchronous optical network (SONET) transmission, the 64 kb/s circuit-switching paradigm continued as the main technology. However, the transition from circuit to cell switching — that is, the asynchronous transfer mode (ATM) network — has completely changed the problems and solutions of telephone networks. Now, telephone networks are very close to computer networks. A modern switching node in telephone networks is almost a general-purpose computer equipped with high fault-tolerance capability and powerful I/O capability. A mesh-like network topology is being used instead of a fully connected topology. Techniques for telephone services resemble those for real-time communication services in packet-switched computer networks, in that both services rely on similar principles such as dedicated resources and static routing. Therefore, the dependability techniques of telephone networks are worth a close look. While telephone network survivability is accounted for at various levels, here we cover only the network-layer operation, which is most relevant to this article.

Essentially, when a telephone connection is broken it is rerouted by detouring the failure point. Failure recovery should be fast so that users (or applications) hardly notice the service disruption caused by the failure. Even more important is to ensure the success of failure recovery itself. If there are not enough resources available for rerouting all affected connections, some of them should be dropped. To avoid resource shortage during failure recovery, *spare resources* (i.e., cold-standby resources) are reserved in advance. The amount of spare resources to be allocated is an important design issue, and is closely related to the problem of selecting the rerouting paths of failed connections. For rerouting, there are three strategies: *local rerouting*, *local-to-end rerouting*, and *end-to-end rerouting*. Each of these strategies is illustrated in Fig. 2.

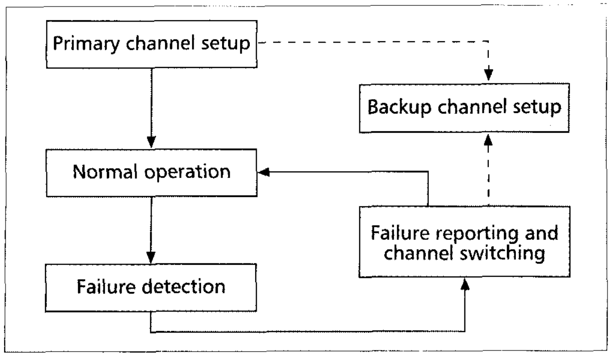
The local-rerouting strategy, also called a *span-restoration* method, has usually been used in synchronous transfer mode (STM) networks. In most of the work on this strategy [12, 13], a "maximum flow" model is used to find the (semi-) optimal placement of spare resources under a deterministic failure hypothesis, typically a single-link failure model. A drawback of the local rerouting approach is that resource usage becomes inefficient after failure recovery, because channel paths tend to be lengthened by local detouring. According to [14], end-to-end rerouting is the best with respect to resource efficiency, local-to-end rerouting is the second, and local rerouting is the worst.

The end-to-end rerouting strategy, also called *path restoration*, has been studied mainly in the context of ATM networks. Essentially, this strategy requires the intervention of the endpoints of each failed connection. There are two variations in this strategy, depending on whether the failure recovery paths are precomputed before failure occurrence, or determined after failures actually do occur. There are several differences between these two methods. In the former the prerouted recovery paths should be disjoint with the corresponding original connection paths, while in the latter the recovery paths can use the healthy components of their original connection paths. The greater flexibility in routing the recovery paths implies the potential of higher resource efficiency of the latter. However, the former has an advantage over the latter in terms of dependability guarantees. In the former, since the network can reserve the spare resources necessary for the recovery connections whose paths are already decided, there will be no conflict/contention for spare resources between recovery attempts upon failures. By contrast, in the latter, when failures occur each disabled connection will try to establish a new connection by "claiming" the shared spare resources. Therefore, resource contention similar to the reactive method can happen, and some connections may need to try several recovery paths before they succeed. Some recent efforts on the former method can be found in [14–16], and an example of the latter method is [17]. Essentially, they try to optimally route recovery paths by minimizing the spare resource reservation, while guaranteeing successful recovery under a deterministic failure model.

Comparison

Basically, our approach uses end-to-end rerouting with precomputed recovery paths. We set up one or more *backup channels* in advance in addition to each *primary channel*; that is, each dependable real-time ($\mathcal{D}$) connection consists of one primary channel and backup channels. Upon failure of a primary channel, one of its backups is promoted to a new primary channel.

In Table 1 existing approaches and our approach are compared with each other in terms of resource overhead, recovery delay, and recovery guarantee. The SFI method is similar to the span restoration method in many aspects, except that it induces higher overhead, because in the SFI method all the spare resources required by each connection are reserved, unlike the span restoration method which optimizes the allocation of spare resources. Both methods will have a shorter recovery delay than the end-to-end rerouting methods, because failures are handled locally without intervention of end nodes. The recovery delay of our approach is smaller than that of the reactive method, because in our approach a backup channel is established before failures actually occur and the network can use it immediately upon the failure of its primary



■ Figure 3. *An overview of our failure-recovery scenario.*

channel, without the time-consuming channel (re)establishment process.[2]

The path restoration method for ATM networks comes closet to our approach, but there are three main differences between the two. First, the path restoration method is unable to control the fault-tolerance level of each connection, and all connections are treated equally under the same failure model. By contrast, our approach allows for per-connection fault-tolerance control, so more critical connections will get higher levels of fault tolerance. Second, in path restoration all channel paths and spare resources are determined simultaneously; hence, addition or removal of a channel requires recalculation of all channel paths and spare resources. It is assumed that the connection demands — virtual path (VP) setup requests — are known at the time of network design and change very rarely.[3] Therefore, this method cannot be applied to an environment where short-lived channels are set up and torn down frequently. In contrast, our approach needs only information that can easily be obtained at runtime. It is possible because we separate the spare resource allocation problem from the channel routing problem, so:

- A backup path may be selected by any algorithm.
- Spare resource allocation may be done with given routing results.

Third, although the control of recovery procedures might be distributed, centralized, or a hybrid of the two, calculation/assignment of spare resources is usually centralized in existing path restoration methods. In our approach, both runtime failure recovery and spare resource allocation are done in a fully distributed manner.

Figure 3 gives an overview of our failure-recovery scenario. The key steps in our approach are:

- Backup channel establishment
- Failure detection
- Failure reporting and channel switching
- Resource reconfiguration

The rest of this article will outline each of these steps.

Connection Establishment

A backup channel does not carry any data until it is activated, so it does not consume bandwidth in a normal situation. However, a backup channel is not free, since it requires the same amount of resources to be reserved as its primary channel in order to provide the same QoS upon its activation. In a normal situation, spare resources can be used by non-real-time traffic, but they cannot be used to set up other real-time channels. This is because if a real-time channel is established by using spare resources which are reserved for other backups, its QoS guarantee may be violated when spare resources are claimed for failure recovery. As a result, equipping each $\mathcal{D}$ connection with a single backup routed disjointly with its primary reduces network capacity by 50 percent or more (because the backup is likely to run over a longer path). Thus, raw backup channels are too expensive to be useful for multimedia networking.

To alleviate this problem we have developed a resource-sharing technique, called *backup multiplexing*. Its basic idea is that on each link, we reserve only a very small fraction of link resources needed for all backup channels going through the link. In this article we consider only link bandwidth for simplicity, but other resources like buffers and CPUs can be treated similarly.

With backup multiplexing, backup channels are overbooked via a *meta-admission test*, in which some existing backup channels are not accounted for in the admission test of a new backup channel. It is, in essence, equivalent to resource sharing between the new backup and those backups unaccounted for. Deciding which backup channels will not be accounted for in the admission test of a backup channel is a crucial problem. Our strategy is to multiplex those backups which are less likely to be activated simultaneously. The probability of simultaneously activating two different $\mathcal{D}$ connections' backups, B_i and B_j, is equal to — actually, bounded by — the probability of simultaneous failures of their respective primary channels. Let's denote this probability $\mathcal{S}(B_i, B_j)$. Based on this probability, the set of backups to be multiplexed together is determined for each backup on a link (i.e., hop-by-hop multiplexing). B_i and B_j are multiplexed if $\mathcal{S}(B_i, B_j)$ is smaller than a certain threshold ν, called the *multiplexing degree*, which is specific to each backup. More accurately, if $\mathcal{S}(B_i, B_j) < \nu_i$, B_j can be multiplexed with B_i. Each backup can use a different ν to determine the set of backups to be multiplexed. The smaller the ν of a backup, the higher fault tolerance will result, since fewer backups will be multiplexed with it. This way, more important connections can have higher fault tolerance (i.e., tolerate harsher failures). However, we require each backup to have the same multiplexing degree on all of its links for easier management.

Figure 4 describes an algorithm that calculates the amount of spare resources (s_ℓ) at link ℓ when a new backup B_i is established on ℓ. Let $\Pi_{B_{i,\ell}} = \{B_\alpha, B_\beta, \ldots\}$ denote the set of backups which are *not* multiplexed with B_i on ℓ, and r_k is the resource requirement of B_k. After calculating all $\Pi_{B_{i,\ell}}$, the highest resource requirement among all sets of $\{\Pi_{B_{i,\ell}} + B_i\}$ is chosen as s_ℓ. To construct $\Pi_{B_{i,\ell}}$, we consider only backups with no multiplexing degrees greater than that of B_i. This is to avoid the risk of overestimating the spare resource requirement by accounting for all backups regardless of multiplexing degree.

In establishing a $\mathcal{D}$ connection, we consider two key dependability-QoS parameters (P_r, Γ), P_r is the probability of fast failure recovery, and Γ is the estimated failure-recovery delay. In other words, the probability that a $\mathcal{D}$ connection will suffer a service disruption longer than Γ is not greater than P_r. Because fast failure recovery relies on the availability of backups, P_r of a $\mathcal{D}$ connection increases with the number of its backups. Backup multiplexing also affects P_r by reducing the amount of spare resources by reserving fewer resources than the sum of resources needed by individual backups, but creat-

[2] *The time required to establish a real-time channel is relatively large and can be unbounded even without contention, since channel establishment requests are usually sent as best-effort messages and the admission test is required at each component on the channel path.*

[3] *In telephone ATM networks, each call setup is handled at the virtual connection (VC) level without requiring a new VP to be set up.*

ing a possibility of spare resource exhaustion. Thus, some backups cannot be activated because other activated backups have already taken all spare resources. In such a case, a *multiplexing failure* is said to occur. We account for the probability that a backup suffers a multiplexing failure in the calculation of P_r. See [18] for a detailed account of calculating P_r.

Thus far, we have explained the procedure of allocating spare resources, assuming that backup paths are given. How to route backups greatly influences the resource efficiency, that is, the amount of spare resources to achieve a certain P_r. Our study shows that traditional routing algorithms such as minimum-hop or maximum load-balanced routing are less effective in backup routing than routing methods which capitalize on the characteristics of backup multiplexing.

```
loop for each backup channel $B_k$ on link $\ell$
    if $S(B_k, B_i) \ge v_i$ and $v_k \le v_i$ then
        $\Pi_{B_i,\ell} \leftarrow \{\Pi_{B_i,\ell} + B_k\}$
    endif
endloop
$s_\ell \leftarrow max\{(\Sigma_k^{\Pi_{B_j,\ell}} r_k) + r_j\}, \forall B_j \in \ell$
```

■ Figure 4. *The backup multiplexing algorithm.*

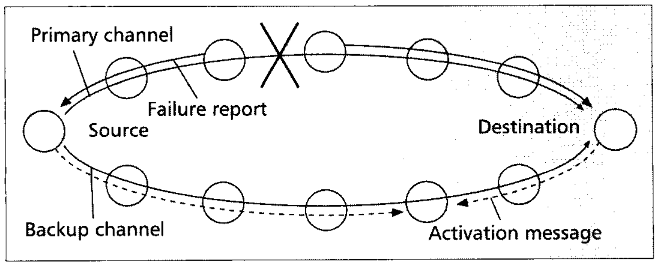
Failure Detection

Effective failure detection with high coverage and low latency is essential for fast failure recovery. Instead of adopting expensive failure-detection techniques (e.g., hardware duplication/comparison for telephone network switches [19], we use behavior-based detection techniques that do not require any special hardware support and hence can be used in *any* network. They are the end-to-end and neighbor detection methods.

End-to-end detection involves both the source and destination nodes of a real-time channel. The source node regularly injects "channel heartbeats" into the channel message stream. A channel heartbeat is a sort of real-time message, and the intermediate nodes on a channel do not discriminate channel heartbeats from data messages. Each channel heartbeat contains the sequence number of the latest data message. In this way, the destination node can monitor the number of data messages lost. If the message loss rate of the channel exceeds the channel's threshold, the destination node declares that the channel has failed. This method will uncover *all* channel failures (i.e., 100 percent failure detection coverage). However, its detection latency is tightly coupled with the application-specific failure threshold, so applications with large failure thresholds will result in long detection latencies.

Neighbor detection resembles the gateway failure-detection protocols in the Internet. Adjacent nodes periodically exchange node heartbeats ("I am alive"). If a node does not receive heartbeats from one of its neighbors for a certain period, it declares all the channels going through the silent neighbor as failed. This scheme has a much weaker dependency on the channel-specific characteristics than the end-to-end scheme, because it monitors the behavior of a neighbor node rather than that of a particular real-time channel. If a node crashes quickly after failures occur, one can achieve very small detection latencies with this scheme. However, there is a possibility that a node does not correctly transmit data messages, but still generates node heartbeats for a while. In such a case, the neighbor scheme may miss the failures of some channels or detect them later than the end-to-end scheme.

We experimentally evaluated the effectiveness of the two failure-detection schemes. The experimental results on a laboratory testbed which was designed without any particular consideration for fault tolerance have indicated that the neighbor scheme detects a significant portion of failures very quickly, while a long latency occasionally results. (Detailed experimental results are reported in [20].)



■ Figure 5. *An example of failure reporting and backup activation.*

Failure Reporting and Channel Switching

If the node that detects the failure of a channel is different from the node responsible for channel switching, the detected failure should be reported to the latter node. There are three important issues in reporting failures. First, who needs to receive failure reports? Second, which path will be used for reporting failures? Third, what information needs to be carried in a failure report? Our approach to these issues is:

- Failure reports are sent from the failure-detecting nodes to the end nodes of failed channels.
- Failure reports are delivered through healthy segments of the failed channels' paths.
- Each failure report contains the channel-id of the failure channel.

Our approach handles multiple (near-) simultaneous failures very naturally and easily. A failure report will be discarded by a node when the failure report about the same channel has already been received by, or passed through, the node. Thus, if multiple failures occur to a channel, only one failure report will reach its end nodes; all the other reports will be lost due to the failures themselves or discarded by intermediate nodes. If the spare resources at a link are exhausted by the activation of backups, the remaining backup channels on the link cannot function as standby channels (i.e., multiplexing failures). Multiplexing failures should be reported in the same way as component failures.

When an end node of a $\mathcal{D}$ connection receives a failure report on its primary channel, it selects one of its healthy backups[4] and sends an *activation message* along the path of the selected backup. (Both end nodes may start backup activation simultaneously.) If an activation message reaches a node on which the backup channel has already been activated by the activation message from the other end node, the activation message is discarded by the node. Figure 5 illustrates the channel-switching procedure described above. If a failure is detected by the channel's destination node (e.g., with the end-to-end method), there will be no explicit failure reporting; instead, the destination node will immediately start the backup activation procedure. To minimize service disruption, data transfer through the new primary channel will be resumed immediately after the source node sends an activation message or receives an activation message generated by the destination node. Albeit unlike-

[4] *The health of backups should also be monitored.*

ly, if a data message arrives at intermediate nodes of the new primary channel before the channel is activated, the data message will be discarded with no harm.

The transmission of failure reports and activation messages is time-critical, because their delays directly affect the service disruption time. The delay of such messages is unpredictable if transported as best-effort messages. Assigning the highest priority to such messages is not a good solution either, since it may affect the QoS of regular real-time traffic. Suppose there are malicious nodes or a large number of coincident failures. In such cases, flooding urgent messages can paralyze the whole (or part of the) network. To achieve delay-bounded and robust transmission of time-critical control messages, we transmit them over special-purpose real-time channels, called *real-time control channels* (RCCs). When the network is initialized, a pair of RCCs, one in each direction, is established on every link of the network. If the capacity of the RCC on each link is large enough to accommodate all time-critical control messages on the link, timely delivery of such messages can be guaranteed.

	Case 1	Case 2	Case 3
Primary channel	x	O	x
Backup channel	x	x	O

■ Table 2. *Cases requiring resource reconfiguration.*

Resource Reconfiguration

In a normal situation, the dependability QoS of a $\mathcal{D}$ connection is maintained by limiting the admission of new connections not to impair the QoS of existing connections. Upon occurrence of a failure, more explicit actions (i.e., resource reconfiguration) need to be taken to preserve the QoS of the connections which are directly or indirectly affected by the failure. Specifically, there are three cases which require resource reconfiguration for QoS maintenance, as shown in Table 2, where x denotes failure and O nonfailure (thus healthy).

In case 1, both primary and backup channels will be reestablished. In case 2, the faulty backup will be torn down and a new backup should be established to maintain the dependability QoS of the injured connection. The network should tear down the old backup before a new backup is established so that the new backup can be routed using the healthy components on the old backup path, if necessary. In case 3, the healthy backup will be promoted to a new primary channel, where the faulty primary will be torn down. After channel switching a new backup is necessary, since the original backup has been activated, thus ceasing its backup role.

Even when a connection is not directly inflicted with failures, its dependability QoS can be affected by the failure recovery for other connections. This is because spare resources are shared by multiple backups, and activation of a backup will reduce the spare resources on its path, and as a result the remaining backups on this path may not receive their original QoS.[5] At such links, more spare resources have to be allocated to maintain the same QoS for the remaining backups. Here the network has to take care of a situation where there are not enough resources available at a link to match the need for additional spare resources. The network can resolve such situations by moving some of the remaining backups to different paths or by QoS degradation. In this article we will not address this issue any further.

Performance Analysis

We evaluated the resource overhead and fault-tolerance capability of our scheme through simulations. The simulated networks were an 8 x 8 torus (wrapped mesh) and an 8 x 8 mesh. To obtain a similar total capacity for both networks, we set the link capacity of the torus to 200 Mb/s and that of the mesh to 300 Mb/s. At each simulation run, a total of 4032 connections were established so that the source and destination of each connection were evenly distributed. Channels of each $\mathcal{D}$ connection were routed disjointly with shortest-path routing. For simplicity, we assumed that all channels used the same traffic model so each channel required 1 Mb/s of bandwidth on each link of its path. We also assumed that the same multiplexing degree was used for all backup channels. Three failure models (i.e., single link failure, single node failure, and double node failures) were simulated. As a metric of the fault-tolerance level achieved by each backup configuration, the ratio of fast recovery to the number of failed primary channels was used. For instance, a 90 percent fast recovery ratio means that 90 percent of the $\mathcal{D}$ connections whose primary failed were recovered by using their backup channels.

The simulation results are summarized in Table 3. The fast recovery ratio given in Table 3 is the average value collected from all possible failure cases under the corresponding failure model. The spare bandwidth value is the average

	$\nu = 0$	$\nu = \lambda$	$\nu = 3\lambda$	$\nu = 5\lambda$	$\nu = 6\lambda$
(a) Single backup in 8 x 8 torus					
Spare bandwidth	35%	30.25%	22.5%	16%	9.5%
One link failure	100%	100%	100%	97.27%	74.11%
One node failure	100%	100%	100%	89.99%	64.72%
Two node failures	93.11%	93.11%	92.98%	84.05%	58.36%
(b) Double backups in 8 x 8 torus					
Spare bandwidth	N/A	N/A	30.25%	21.25%	12.88%
One link failure	N/A	N/A	100%	100%	100%
One node failure	N/A	N/A	100%	100%	97.68%
Two node failures	N/A	N/A	100%	99.82%	93.28%
(c) Single backup in 8 x 8 mesh					
Spare bandwidth	35.47%	33.11%	24.47%	19.69%	17.22%
One link failure	100%	100%	100%	97.63%	90.39%
One node failure	100%	100%	99.94%	91.74%	84.08%
Two node failures	89.11%	89.22%	88.83%	81.82%	75.32%

■ Table 3. *Simulation results (λ = component failure rate).*

[5] *In the worst case, the remaining backups can be subject to exhaustion of spare resources.*

[6] *$\nu = 0$ has the same effect as disabling backup multiplexing, since no backup will be multiplexed with each other.*

of all links. N/A indicates that the total bandwidth requirement exceeded the network capacity before establishing all connections.

In general, with far less than half the resource overhead as that before multiplexing,[6] more than 90 percent of fault-tolerance capability could be preserved by backup multiplexing in the single-backup configuration. As expected, the use of a smaller multiplexing degree results in higher fault tolerance with an exception between $\nu = 0$ and $\nu = \lambda$ in the bottom row of Table 3c, where the fault-tolerance level was increased after backup multiplexing. This is due to the impact of backup multiplexing on the channel route selection. The multiplexing efficiency was even further improved by using the double backup configuration. Thus, a similar level of fault tolerance was achievable with significantly less spare resources in the double backup configuration. For example, compare the case of single backup with $\nu = 3\lambda$ with the case of double backups with $\nu = 6\lambda$ in the torus network. Another interesting observation is that multiplexing efficiency is affected by network topology. In the mesh network, the reduction of spare resources by multiplexing is not as great as in the torus network. This is because the absence of wrapped links in the mesh network makes the primary channel paths more concentrated on the central region of the network, thus discouraging multiplexing among their backups.

Concluding Remarks

The main focus of this article is twofold. First, we surveyed existing approaches to dependable multimedia communication and discussed their pros and cons. Second, we presented a new scheme which selectively adopts only the merits of existing approaches without their shortcomings. Thus, this scheme allows the desired level of fault tolerance to be achieved for each connection at an acceptably low level of resource overhead.

This scheme scales well because it does not require each node to maintain global knowledge of the network traffic conditions or to generate any type of messages to be broadcast. Control messages are sent only over those paths of channels affected by failures. Backup multiplexing is performed hop by hop; therefore, at each link only the knowledge of primary channels whose backups traverse the link is required. Such information can easily be collected by making a backup channel establishment message carry the path information of its primary channel. Since our scheme is designed without any assumption for a particular real-time communication scheme, it can be placed on top of any existing (possibly independently developed) real-time channel protocols.

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