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An Image Transport Protocol for the Internet

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Abstract

Images account for a significant and growing fraction of Web downloads. The traditional approach to transporting images uses TCP, which provides a generic reliable, in-order byte-stream abstraction, but which is overly restrictive for image data. We analyze the progression of image quality at the receiver with time and show that the in-order delivery abstraction provided by a TCP-based approach prevents the receiver application from processing and rendering portions of an image when they actually arrive. The end result is that an image is rendered in bursts interspersed with long idle times rather than smoothly.

This paper describes the design, implementation, and evaluation of the Image Transport Protocol (ITP) for image transmission over loss-prone congested or wireless networks. ITP improves user-perceived latency using application-level framing (ALF) and out-of-order Application Data Unit (ADU) delivery, achieving significantly better interactive performance as measured by the evolution of peak signal-to-noise ratio (PSNR) with time at the receiver. ITP runs over UDP, incorporates receiver-driven selective reliability, uses the Congestion Manager (CM) to adapt to network congestion, and is customizable for specific image formats (e.g., JPEG and JPEG2000). ITP enables a variety of new receiver post-processing algorithms such as error concealment that further improve the interactivity and responsiveness of reconstructed images. Performance experiments using our implementation across a variety of loss conditions demonstrate the benefits of ITP in improving the interactivity of image downloads at the receiver. Even though we explore image transport, ITP is a generic selectively reliable unicast transport protocol with congestion control that can be customized for specific applications and formats.

1 Introduction

Images constitute a significant fraction of traffic on the World Wide Web, e.g., according to a recent study, JPEGs account for 31% of bytes transferred and 16% of documents downloaded in a client trace [15]. The ability to transfer and render images on screen in a timely fashion is an important consideration for content providers and server operators because users surfing the Web care about interactive latency. At the same time, download latency must be minimized without compromising end-to-end congestion control, since congestion control is vital to maintaining the long-term stability of the Internet infrastructure. In addition, appropriate reaction to network congestion also allows image applications to adapt well to available network conditions.

The HyperText Transport Protocol (HTTP) [11] uses the Transmission Control Protocol (TCP) [31] to transmit images on the Web. While the use of TCP achieves both reliable data delivery and good congestion control, these come at a cost—interactive latency is often significantly large and leads to images being rendered in "fits and starts" rather than in a smooth way. The reason for this is that TCP is ill-suited to transporting latency-sensitive images over loss-prone networks where losses occur because of congestion or packet corruption. When one or more segments in a window of transmitted data are lost in TCP, later segments often arrive out-of-order at the receiver. In general, these segments correspond to portions of an image that may be handled upon arrival by the application, but the in-order delivery abstraction imposed by TCP holds up the delivery of these out-of-order segments to the *application* until the earlier lost segments are recovered. As a result, the image decoder at the receiver cannot process information even though it is available at the lower transport layer. The image is therefore rendered in bursts interspersed with long delays rather than smoothly.

The TCP-like in-order delivery abstraction is appropriate for image encodings in which incoming data at the receiver can only be handled in the order it was transmitted by the sender. Some compression formats are indeed constrained in this manner, e.g., the Graphical Interchange Format, GIF [14] which uses lossless LZW compression [21, 43] on the entire image. However, while some compression formats are constrained in this manner, several others are not. Notable examples of formats that encourage out-of-order receiver processing include JPEG [42, 29] and the emerging JPEG2000 standard [20]. In these cases, a transport protocol that facilitates out-of-order data delivery allows the application to process and render portions of an image as they arrive, improving the interactivity and perceived responsiveness of image downloads. Such a protocol also enables the image decoder at the receiver to implement effective error concealment algorithms on partially received portions of an image, further improving perceived quality.

One commonly suggested approach to tackling this problem of in-order delivery is to extend existing TCP implementations and its application programming interface so that received data can be consumed out-of-order by the application. However, merely tweaking an in-order byte-stream protocol like TCP without any additional machinery to achieve the desired effect is not adequate because out of order TCP segments received by the application in this man-

ner do not correspond in any meaningful way to processible data units at the application level.

We propose the Image Transport Protocol (ITP), a transport protocol in which application data unit (ADU) boundaries are exposed to the transport module, making it possible to perform out-of-order delivery. Because the transport is aware of application framing boundaries, our approach expands on the application-level framing (ALF) philosophy, which proposes a one-to-one mapping from an ADU to a network packet or protocol data unit (PDU) [9]. However, ITP deviates from the TCP-like notion of reliable delivery and instead incorporates selective reliability, where the receiver is in control of deciding what is transmitted from the sender at any instant.

This form of reliability is appropriate for heterogeneous network environments that will include a wide variety of clients with a large diversity in processing power, and allows the client, depending on its computational power and available suite of image decoding algorithms, to request application data that would benefit it the most. Furthermore, other image standards such as JPEG2000 support region-of-interest (ROI) coding that allows receivers to select portions of an image to be coded and rendered with higher fidelity. Receiver-driven selective reliability is important if applications are to benefit from this feature.

Despite the disadvantages of in-order delivery as far as interactivity is concerned, using TCP has significant advantages from the viewpoint of congestion control. Any deployable transport protocol must perform congestion control for the Internet to remain stable, which suggests that a significant amount of additional complexity would have to be designed and implemented in ITP. Fortunately, we are able to leverage the recently proposed Congestion Manager (CM) [2, 3] to perform stable, end-to-end congestion control.

In this paper, we describe the motivation, design, implementation, and evaluation of ITP, an ALF-based image transport protocol. Our key contributions are as follows.

- We present the design of ITP, a transport protocol that runs over UDP, incorporating out-of-order data delivery and receiver-controlled selective reliability. We have designed ITP so that it can be used with no modifications to higher layer protocols such as HTTP [4, 11] or FTP [32].
- We show how to tailor ITP for JPEG image transport, by introducing a framing strategy and tailoring the reliability protocol by scheduling request retransmissions.
- ITP's out-of-order delivery enables many receiver optimizations. We describe one such optimization in which missing portions of an image are interpolated using a simple error concealment algorithm.
- We present the measured performance of a user-level implementation of ITP across a range of network conditions that demonstrate that the rate of increase in PSNR with time is significantly higher for ITP compared to TCP-like in-order delivery of JPEG images.

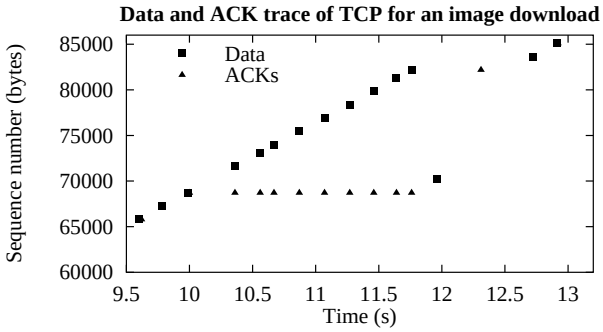

Figure 1. Portion of packet sequence trace of a TCP transfer of an image.

The remainder of this paper is organized as follows. In Section 2, we present empirical evidence in favor of our approach and discuss our design goals for ITP. Section 3 describes various aspects of the ITP protocol — out-of-order delivery, receiver-reliability, and congestion management. This is followed by a discussion on applying ITP to JPEG transport in Section 4. In Section 5, we present the measured performance of ITP that demonstrates the advantages over the traditional TCP approach under a variety of conditions. Finally, we discuss related work in Section 6 and conclude in Section 7.

2 Design considerations

We start by motivating our approach by highlighting the disadvantages of using TCP. The main drawback of using TCP for image downloads is that its in-order delivery model interferes with interactivity. To demonstrate this, we conducted an experiment across a twenty-hop Internet path to download a 140 KByte image using HTTP 1.1 running over TCP. The loss rate experienced by this connection was 2.3%, 3 segments were lost during the entire transfer, and there were no sender retransmission timeouts.

Figure 1 shows a portion of the packet sequence trace obtained using `tcpdump` running at the receiver. We see a transmission window in which exactly one segment was lost, and all subsequent segments were received, causing the receiver to generate a sequence of duplicate ACKs. There were ten out-of-sequence segments received in all waiting in the TCP socket buffer, none of which was delivered to the image decoder application until the lost segment was received via a (fast) retransmission almost 2.2 seconds after the loss. During this time, the user saw no progress, but a discontinuous spurt occurred once this lost segment was retransmitted to the receiver, and several kilobytes worth of image data were passed up to the application.

To understand how ordering semantics influence the perceptual quality of the image, we conduct a second experiment where the image is downloaded over TCP and study the evolution of image "quality", as measured by peak signal-to-

noise ratio (PSNR) [36] with respect to the original transmitted image. Figure 2 shows this for a transfer that experiences a 15% loss rate. We find that the quality remains unchanged for most of the transfer, due to an early segment loss, but rapidly rises upon recovery of that lost segment. A smoother evolution in PSNR, as in the "ideal" transfer which does out-of-order delivery is desirable for better interactivity.

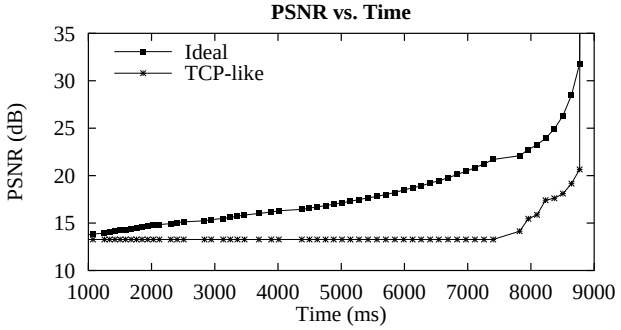

Figure 2. PSNR evolution of the rendered image at the receiver for a TCP transfer with 15% loss rate.

We observe that a design in which the underlying transport protocol delivers out-of-sequence data to the application might avoid the perceived latency buildup. In order to do this, the transport "layer" (or module) must be made aware of the application framing boundaries, such that each data unit is independently processible by the receiver.

In this section, we discuss the key considerations that directed the design of ITP.

1. *Support out-of-order delivery of ADUs to the application, while efficiently accommodating ADUs larger than a PDU.*
   Our first requirement is that the protocol accommodate out-of-order delivery, but does so in a way that allows the receiver application to make sense of the mis-ordered data units it receives. In the pure ALF model [9], each ADU is matched to the size of a protocol data unit (PDU) used by the transport protocol. This implies that there is no "coupling" between two packets and that they can be processed in any order. Unfortunately, it is difficult to ensure that an ADU is always well matched to a PDU because the former depends on the convenience of the application designer and what is meaningful to the application, while the latter should not be too much larger (if at all) than the largest datagram that can be sent unfragmented, in order to minimize retransmission overhead in the event of a packet loss. This means that there are times when an ADU is larger than a PDU, requiring an ADU to be fragmented by the transport protocol for efficiency.

2. *Support receiver-controlled selective reliability.*
   Our next design consideration addresses reliability. When packets are lost, there are two possible ways of handling retransmissions. The conventional approach is for the sender to detect losses and retransmit them in the order in which they were detected. While this works well for protocols like TCP that simply deliver all the data sequentially to a receiver, interactive image transfers are better served by a protocol that allows the receiving application (and perhaps user) to exercise control over how retransmissions from the sender occur. For example, a user should be able to express interest in a particular region of an image, causing the transport protocol to prioritize the transmission of the corresponding data over others. In general, the receiver knows best what data it needs, if any, and therefore allowing it to control requests for retransmission is best-suited to improving user-perceived quality.

3. *Support easy customization for different image formats.*
   This design consideration is motivated by the observation that there are many different image formats that can benefit from out-of-order processing, each of which may embed format-specific information in the protocol. For example, the JPEG format uses an optional special delimiter called a *restart marker*, which signifies the start of an independently processible unit to the decoder by resynchronizing the decoder. Such format- or application-specific information should be made available to the receiver in a suitable way, without sacrificing generality in the basic protocol.

   The design of ITP borrows from from the design of other application-level transport protocols such as the Real-time Transport Protocol (RTP) [37]. In ITP, as in RTP, a base header is customized by individual application protocols, with profile-specific extension headers incorporating additional information.

4. *Application and higher-layer protocol independence.*
   While this work is motivated by interactive image downloads on the Web, we do not want to restrict our solution to just HTTP. In particular, we do not require any changes to the HTTP specification but rather replace TCP with ITP at the transport layer. Since ITP provides loss recovery, we use a duplex ITP connection to carry HTTP request messages such as GET and PUT as well as HTTP responses.

5. *Sound congestion control.*
   Finally, congestion-controlled transmissions are important for deploying any transport protocol on the Internet. But rather than reinvent complex machinery for congestion management (a look at many of the subtle bugs in TCP congestion control implementations that researchers have discovered over the years shows that this is not straightforward [27]), we leverage the recently developed Congestion Manager (CM) architecture [2]. The CM abstracts away all congestion control into a trusted kernel module independent of transport protocol, and provides a general API for applications to learn about and adapt to changing network conditions [3]. Our design uses the CM to perform conges-

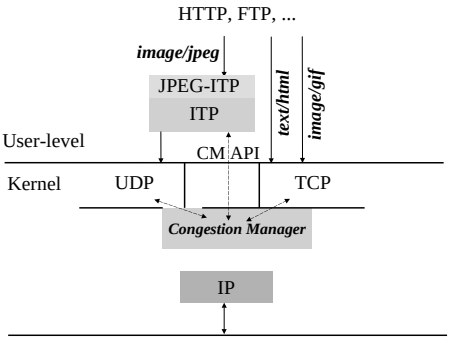

Figure 3. The system architecture showing ITP, its customization for JPEG, and how HTTP uses it instead of TCP for MIME type "image/jpeg" while using a conventional TCP transport for other data types. All HTTP protocol messages are sent over ITP, not just the actual image data, which means that ITP replaces TCP as the transport protocol for this data type.

tion control, with packet transmissions orchestrated by the CM via its API.

3 ITP Design

In this section, we describe the internal architecture of ITP and the techniques used to meet the aforementioned design goals. ITP is designed as a modular user-level library that is linked by the sender and receiver application. The overall system architecture is shown in Figure 3, which includes an example of an application protocol such as HTTP or FTP using ITP for data with MIME type "image/jpeg" and TCP for other data. It is important to note that ITP "slides in" to replace TCP in a way that requires no change to the specification of a higher-layer protocol like HTTP or FTP. A browser initiates an ITP connection in place of a TCP connection if a JPEG image is to be transferred. The HTTP server initiates an active open on UDP port 80 and aaits client requests that are made using the HTTP/ITP/UDP protocol.

3.1 Out-of-order Delivery

ITP provides an out-of-order delivery abstraction. Providing such an abstraction at the granularity of a byte, however, makes it hard for the application to infer what application data units a random incoming sequence of bytes corresponds to. The application handles data in granularities of an ADU, so ITP provides an API by which an application can send or receive a complete ADU. We now describe the mechanics of data transfer at the sending and receiving ITP hosts.

The sending application invokes `itp_send()` to send an ADU to the receiver. Before shipping the ADU, ITP incorporates a header, shown in Figure 4 that includes an incrementing ADU sequence number and ADU length. The sequence number and length of an ADU are used by the receiver to detect the loss of an ADU or the loss of a sequence of bytes within the ADU, perform reassembly within an ADU, and verify that the complete ADU has arrived.

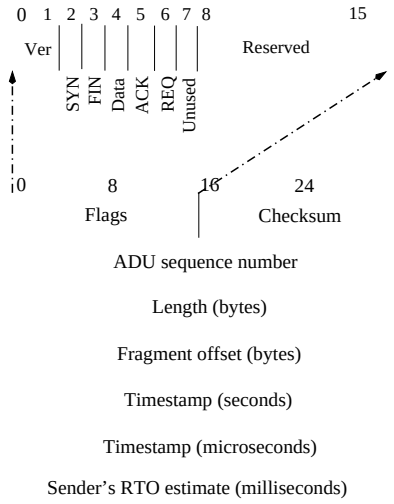

Figure 4. The 28-byte generic ITP transport header contains meta-data pertaining to each fragment, as well as the ADU that the fragment belongs to, such as the ADU sequence number and length, the fragment offset within the ADU, a sender timestamp, and the sender's estimate of the retransmission timeout.

When a complete ADU arrives at the receiver, the ITP receiver invokes a well-known callback function implemented by the application, called `itp_app_notify()`. In response, the application calls an ITP library function `itp_read()` to read the incoming ADU into its own buffers, and returns control to ITP. This interaction is shown in Figure 5. The important point to note is that this sequence of steps occurs when a complete ADU arrives at the receiver, *independent* of the order in which it was transmitted from the sender.

Unfortunately, not all ADUs are small enough to fit in one PDU which is the maximum unfragmented datagram

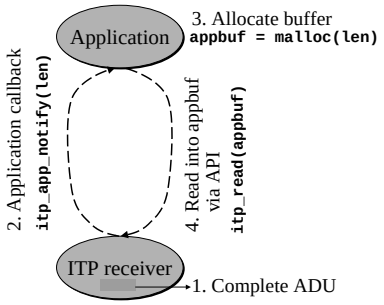

Figure 5. The sequence of operations when a complete ADU arrives at the ITP receiver.

on the path to the receiver. This requires that any ADU larger than a PDU be fragmented into PDU-sized units before transmission. Using arbitrarily-sized ADUs as the granularity of loss recovery is inefficient. Consider for example an ADU transmitted by the transport protocol that was fragmented by a lower layer for transmission, and exactly one of the fragments was lost in transit. The receiver must ask for the entire ADU to be retransmitted if the unit of naming and transmission by the transport layer is an ADU, thereby degrading protocol goodput. Rather than suffer poor performance caused by redundant retransmissions, ITP bridges the mismatch between network-supported packet sizes and application-defined data units by breaking up an ADU into *fragments* no bigger than the maximum transmission unit of the path and identifying each fragment by its byte-offset and length within an ADU as well as the ADU sequence number.[1] We emphasize that this is done to avoid inefficiencies in retransmission, but is not exposed to the receiving application. As a result, applications are not forced to limit their framing to network packet sizes, and incomplete ADU data are not visible to them.

3.2 Reliability

One of the design goals in ITP is to put the receiver in control of loss recovery which suggests a protocol based on *retransmission request* messages sent from the receiver. In addition to loss recovery, ITP must also reliably handle connection establishment and termination, as well as host failures and subsequent recovery without compromising the integrity of delivered data. We incorporate TCP-like connection establishment and termination mechanisms for this.

3.2.1 Connection management

Although an important application of ITP is downloading images on the Web using HTTP, we do not wish to restrict all higher-layers to HTTP-like protocols where the client initiates the connection. For example, when used by FTP, the server performs the active open rather than the client. We emulate TCP-style connection management in order to support all styles of applications. ITP incorporates the three-way connection establishment procedure of TCP [38]. The initial sequence number chosen by both sides determines the ADU sequence space for each transfer direction[2].

The ITP sender signals the last ADU in a transmission sequence by setting the FIN bit in the flags of the ITP header. The receiver uses this to detect when all transmitted data items (of interest to the receiver) have arrived and to terminate the connection. We use the FIN-ACK mechanism of TCP, transitioning into exactly the same states as a terminating TCP (CLOSE_WAIT for a passive CLOSE; FIN_WAIT_1, optionally followed by CLOSING or FIN_WAIT_2, and then a TIME_WAIT for an active one). As in TCP, the active closer transitions from the TIME_WAIT state to CLOSED after the 2MSL timeout.

This choice of connection establishment and termination procedures allows ITP to handle several different applications (all combinations of servers and clients performing active/passive opens and closes). This decision also allows us to be fairly certain of the correctness of the resulting design.

We do, however, address the significant problem of connections in the TIME_WAIT state at a busy server. The problem is that in most HTTP implementations, the server does the active close rather than the client, which causes the server to expend resources and maintain the TIME_WAIT connections. This design is largely forced by many socket API implementations, which do not allow applications to easily express a half-close.[3] We solve this problem by providing a "half-close" call to the ITP API that allows the client use it. When one side (e.g., an HTTP client, soon after sending a GET message) decides that it has no more data to send, but wants to receive data, it calls `itp_halfclose()` which sends a FIN to the peer. Of course, retransmission requests and data ACKs continue to be sent. In the context of HTTP, the TIME_WAIT state maintenance is therefore shifted to the client, freeing up server resources.

3.2.2 Loss recovery

All retransmissions in ITP occur only upon receipt of a retransmission request from the receiver, which names a requested fragment using its ADU sequence number, fragment offset, and fragment length. While many losses can be detected at the receiver using a data-driven mechanism that observes gaps in the received sequence of ADUs and fragments, not all losses can be detected in this manner. In particular, when the last fragment or "tail" of a burst of fragments transmitted by a sender is lost, a retransmission timer is required. Losses of previous retransmissions similarly require timer-based recovery.

One possible design is for the receiver to perform all data-driven loss recovery, and for the sender to perform all timer-based retransmissions. However, this contradicts our goal of receiver-controlled reliability because the sender has no knowledge of the fragments most useful to the receiver. Unless we incorporate additional complex machinery by which a receiver can explicitly convey this information to the sender, the sender may retransmit old and uninteresting data upon a timeout.

[1] Path MTU discovery [22] can be used to determine this value between a pair of hosts on the Internet.

[2] We do not view the three-way handshake as a performance problem, despite the extra round-trip that it entails; indeed, should this be a concern, it can be modified to allow data to be piggybacked along with the establishment message. We do not recommend this mode as it tends to make defense against denial-of-service attacks like SYN-floods hard to handle (e.g., using SYN-cookies, which ITP can incorporate with little difficulty).

[3] The `shutdown(socket_fd, how)` call, with `how` set to 1 is supposed to cause a half-close, telling the peer that no more data will be originated on the connection, but not all TCP implementations handle this correctly. Furthermore, the Hosts Requirements RFC 1122 lists the half-close as a "MAY implement" option.

Our solution to this problem is to move all timer handling to the receiver. If the receiver detects no activity for a timeout duration, a retransmission request is sent. If no gaps are detected in the received ADU stream, a retransmission request is sent for the next expected ADU, i.e., 1 + last ADU sequence number received, thereby initiating recovery from a tail loss, if one did occur. Since the retransmission timer is always active until a FIN, this message is repeated periodically until the receiver is ready to terminate. However, a half close by a peer causes this timer to be disabled. Note that with this approach, an HTTP server does not end up periodically probing the client requesting more data after a GET request for a URL. Once a half-close is received from the client, the server disables the timer.

It is rather difficult for accurate round-trip time estimation to be performed at the receiver when data flows from sender to receiver. Hence, we allow the sender calculate the retransmission timeout (RTO) as in TCP with the timestamp option [19], and pass this RTO to the receiver in the ITP header (Figure 4).

ITP also incorporates data-driven retransmission requests. To do this, the receiver maintains a maintains a list of incomplete and missing ADUs. When a fragment is received, missing fragments or ADUs are detected by looking up the data structure. The receiver now has three tasks: (i) decide whether it is time to ask for the fragment, (ii) decide how many fragments to ask for, and (iii) if at least one fragment can be requested at this time, decide which fragments to request.

Two considerations dictate whether it is time to ask for a fragment. First, if a request has already been made for the fragment, it should not be made again unless an RTO has elapsed since the first request to allow a retransmitted ADU or fragment to reach the receiver. To do this, the receiver logs the time of last request and ensures that a subsequent request is sent only if the elapsed time is longer than the estimated RTO. Second, packets may get reordered on the Internet [26], and the receiver must guard against asking for a reordered (but not lost) fragment. The approach in TCP is to wait for a threshold number (three) of duplicate ACKs and retransmit the first unacknowledged segment. Unfortunately, this does not work well when windows are small or when ADUs are small in size (as is often the case for ITP applications). Our solution to this problem is motivated by the observation by Paxson that a small delay before sending an ACK in TCP often accounts for reordered segments [28]. ITP modifies this approach by adapting it to the transmission rate r (in fragments/sec) from the sender, which it monitors using an exponentially-weighted moving average filter. The receiver waits for a duration equal to $3/r$ seconds before sending a request, allowing for a typical number of reordered fragments to arrive and cancel a pending retransmission request.

A subtle effect, which is not normally seen in TCP, occurs when receiver-driven selective reliability is implemented in this manner. Consider the case when a timeout occurs and the congestion window at the sender is set to 1, as shown in Figure 6. A retransmission request from the receiver causes the sender to send one requested fragment. When this fragment is ACKed, congestion control causes the sender's window to grow to 2. The sender may have other old data that the receiver has not yet received, but because all reliability is receiver-controlled, the sender cannot unilaterally retransmit old data. The sender therefore decides to send new fragments and use up its newly opened congestion window (slow start), making timely loss recovery of other lost fragments difficult. The receiver therefore loses the ability to order and prioritize a particular set of retransmissions before any other new data is received.

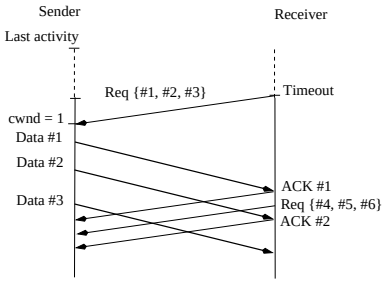

Figure 6. Retransmissions #1, #2, and #3 are transmitted before the next request is received by the sender. Sending three requests in each request message keeps the "pipe" full.

To handle this inversion in transmission priority, the receiver must transmit multiple retransmission requests for if not, the sender either remains idle or continues to send new data on the connection. This problem is solved if the receiver sends at least three retransmission requests each time a loss is detected, assuming that many losses have occurred and the receiver is interested in recovering them. This allows the sender to build up an ordered list of pending retransmissions and use up a newly opened congestion window for retransmissions requested by the receiver rather than new data. Every time a loss is detected, up to three fragments are potentially capable of being transmitted from the sender before the next retransmission request reaches it. The number three is therefore required for efficiency and timely recovery of lost fragments and results from the first two window increments of TCP-style slow start implemented in the CM. Figure 6 illustrates this effect.

The most difficult part in loss recovery is to decide which fragment to request at any time among the missing ones. This is difficult because of the tension between application-specificity and generality. We would like to put the application in control of what to request, but save each application the trouble of writing the complex loss detection code. Furthermore, we would like to provide a reasonable default behavior to handle applications that do not care to customize their reliability schedules.

ITP provides a simple default scheduling algorithm for retransmission requests in which requests are made for fragments from all the missing ADUs starting from the most recent one and progressing in sequence to the least recent, subject to the above conditions of not requesting them too

```
PROCESSRXMITREQ(fragment)
    Send requested fragment via cm_send();
    InformCM();

INFORMCM()
    now ← current_time;
    if (now − last_activity > timeout_duration)
        cm_update(. . ., CM_PERSISTENT, . . .);
    else
        cm_update(. . ., CM_TRANSIENT, . . .);
```

Figure 7. How the ITP sender handles a retransmission request.

soon. More importantly, ITP also allows application-specific customization of reliability, as described in Section 4.2 for JPEG. This allows a JPEG receiver to request only fragments from ADUs that it is currently interested in, based on the decoding algorithm it implements.

We briefly contrast ITP's receiver-driven recovery algorithm with other related approacches such as WebTP. A more detailed comparison of related work is presented in Section 6. ITP's receiver-controlled selective reliability differs in significant ways from WebTP, which does share similar reliability goals. For example, WebTP uses a fully-qualified URL to identify an ADU similar to the work reported in [33], while ITP uses a simpler fixed-length ADU sequence number but disseminates a mapping at the beginning of a connection that enables customization. ITP uses the simpler strategy of sending the RTO in the packet header to the receiver compared to WebTP, which uses estimates the mean inter-arrival packet time[4] and sending a retransmission request if no packet arrived in some deviation from this. ITP incorporates ideas that can be used by a general selectively reliable protocol, but our primary contributions are its customization and evaluation in the context of image transport. The scheduling algorithm presented in Section 4 for JPEG-ITP retransmission requests shows how a receiver can customize the retransmission schedule.

3.3 Using the Congestion Manager

ITP relies on the CM for congestion control, using the CM API to adapt to network conditions and to inform the CM about the status of transmissions and losses [3]. Since ITP reliability is receiver-based, there is no need for positive ACKs from the receiver to the sender for reliability. ACKs from the receiver are solely for congestion control and estimating round-trip times. The CM requires the cooperation of the application in determining the state of the network. By informing the ITP sender about the status of transmissions, an ITP ACK allows the sender to update CM state. When the ITP sender receives an ACK, it calculates how many bytes have cleared the "pipe" and calls `cm_update()` to inform the CM of this.

[4]We believe this is less well-understood than our approach, and note that the congestive collapse episodes of the mid-1980s were largely because of bad retransmission strategies.

When a retransmission request arrives at the sender, the sender infers that packet losses have occurred, attributes them to congestion (as in TCP), and invokes `cm_update()` with the `lossmode` parameter set to CM_TRANSIENT, signifying transient congestion. In a CM-based transport protocol where timeouts occur at the sender, the expected behavior is to use cm_update() with the `lossmode` parameter set to CM_PERSISTENT, signifying persistent congestion. In ITP, the sender never times out, only the receiver does. However, sender sees a request for retransmission transmitted after a timeout at the receiver. Hence, when a retransmission request arrives, the sender determines if the message was transmitted after a timeout or because of out-of-sequence data. We solve this problem by calculating the elapsed time since the last instant there was any activity on the connection from the peer, and if this time is greater than the retransmission timeout value, the CM is informed about persistent congestion. Figure 7 shows the steps taken by an ITP sender when it receives a request for retransmission.

3.4 Design Summary

In summary, ITP provides out-of-order delivery with selective reliability. It handles all combinations of active/passive opens and closes by server and client applications by borrowing TCP's connection management techniques. Application-level protocols like HTTP do not have to change their specifications to use ITP, and ITP is designed to support many different types of applications.

ITP differs from TCP in the following key aspects. It does not force a reliable in-order byte stream delivery and puts the receiver in control of deciding when and what to request from the sender. It uses a callback-based API to deliver out-of-order ADUs to the application. ITP includes a "half-close" method that moves the TIME_WAIT maintenance to the client in the case of HTTP. In TCP the sender detects re-ordered segments only after three duplicate ACKs are received, while in ITP, receivers detect re-ordering based on a measurement of the sending rate. We emphasize that ITP has a modular architecture and relies on CM for congestion control. ACKs in ITP are used solely as feedback messages for congestion control and round-trip time calculation, and *not* for reliability. Rather, receivers control retransmissions by carefully scheduling requests.

4 JPEG Transport using ITP

In this section, we discuss how to tailor ITP for transmitting JPEG images. JPEG was developed in the early 1990s by a committee within the International Telecommunications Union, and has found widespread acceptance for use on the Web. The compression algorithm uses block-wise discrete cosine transform (DCT) operations, quantization, and entropy coding [30]. JPEG-ITP is the customization of ITP by introducing a JPEG-specific framing strategy based on restart markers and tailoring the retransmission protocol by scheduling request retransmissions.

4.1 Framing

The current model for JPEG image transmission on the Internet is to segment it into multiple packets. However, JPEG uses entropy coding, and the resulting compressed bitstream consists of a sequence of variable-length code words. Packet losses often result in catastrophic loss if pieces of the bitstream are missing at the decoder. Arbitrarily breaking an image bitstream into fixed-size ADUs does not work because of dependencies between them. However, JPEG uses *restart markers* to allow decoders to resynchronize when confronted with an ambiguous or corrupted JPEG bitstream, which can result from partial loss of an entropy coded segment of the bitstream. The introduction of restart markers helps localize the effects of the packet loss or error to a specific sub-portion of the rendered image. This segmentation of the bitstream into independent restart intervals also facilitates out-of-order processing by the application layer. The approach used by JPEG to achieve loss resilience provides a natural solution to our framing problem.

When an image is segmented into restart intervals, each restart interval is independently processible by the application and naturally maps to an ADU. The image decoder is able to decode and render those parts of the image for which it receives information without waiting for packets to be delivered in order. The base ITP header is extended with a JPEG-specific header that carries framing information, which includes the spatial position of a 2-byte restart interval identifier.

Our implementation of JPEG-ITP uses 8-bit gray-scale images in the baseline sequential mode of JPEG. We require that the image server store JPEG images with periodic restart markers. This requirement is easy to meet, since a server can easily transcode offline any JPEG image (using the `jpegtran` utility) to obtain a version with markers. When these markers occur at the end of every row of blocks, each restart interval corresponds to a "stripe" of the image. These marker-equipped bistreams produce exactly the same rendered images as the original ones when there are no losses. Since JPEG uses a blocksize of 8x8 pixels, each restart interval represents 8 pixel rows of an image. We use the sequence of bits between two restart markers to define an ADU, since any two of these intervals can be independently decoded. Our placement of restart markers achieves the effect of rendering an image in horizontal rows.

4.2 Scheduling

As discussed in Section 3, ITP allows the application to specify the priorities of different ADUs during recovery. We describe how this is achieved in JPEG-ITP. Figure 8 shows the key interfaces between ITP and JPEG-ITP, and between JPEG-ITP and the decoder. ITP handles all fragments and makes only complete ADUs visible to JPEG-ITP. To preserve its generality, we do not expose application-specific ADU names to ITP. Thus, when a missing ADU needs to be recovered by the decoder, JPEG-ITP needs to map the restart interval number to an ITP ADU sequence number.

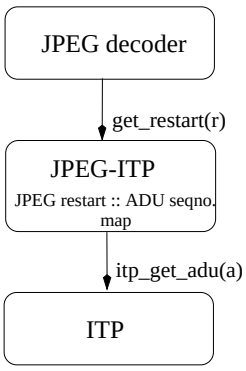

Figure 8. JPEG-ITP maintains a mapping of restart intervals to ADU sequence numbers. The JPEG decoder specifies recovery priorities based on application-level considerations, which is used to guide ITP's request scheduling.

To do this, the JPEG-ITP sender reliably transmits this mapping as the first ADU of the connection, before transmitting the image ADUs. This name map is used to schedule ITP retransmission requests.

ITP maintains a priority list of the retransmission schedule by exporting an asynchronous API function `itp_get_adu()` that customized protocols like JPEG-ITP and applications can use to inform ITP of the desired ADU. ITP uses this priority information to schedule requests for missing fragments from these ADUs ahead of others. In addition, JPEG-ITP exports an API function to the decoder that allows the latter to specify restart intervals that must be prioritized during recovery, e.g., if the decoder uses error concealment as in Section 4.3, this is used to preferentially request ADUs that have not been interpolated from the existing partial image.

4.3 Error Concealment

Out-of-order delivery allows the JPEG decoder to refine a partial image using error concealment based on interpolation techniques. Portions of the image corresponding to the received ADUs are decoded and rendered. Before rendering, a post-processing step is applied to the image to conceal lost stripes. Error concealment exploits spatial redundancy in images and aims to increase the perceptual quality of the rendered image.

Each missing pixel value is the result of a linear interpolation, or average, of its neighbors. This step is applied to all missing restart intervals at the receiver. Therefore, in 2-D, the missing pixel $x_{i,j}$ is given by:

$$x_{i,j} = \frac{x_{i-1,j} + x_{i+1,j} + x_{i,j-1} + x_{i,j+1}}{4} \quad (1)$$

The boundary conditions are determined by the pixel values of neighboring blocks. Using the lexicographic ordering of pixels in a block,

$\mathbf{x} = \{x_{0,0}, x_{0,1}, ...x_{0,B-1}, x_{1,0}..., x_{B-1,B-2}, x_{B-1,B-1}\}$, the estimate of the missing block may be computed as

$$\hat{\mathbf{x}} = A^{-1}\mathbf{c} \tag{2}$$

where A is a block tri-diagonal matrix given by

$$A = \begin{bmatrix} L & I & O & \cdots & \\ I & L & I & O & \cdots \\ O & I & L & I & O \\ \cdots & O & I & L & I \\ & \cdots & O & I & L \end{bmatrix} \tag{3}$$

and L is a 8x8 tri-diagonal matrix formed from $\{1, -4, 1\}$.

$\mathbf{c}$ is a vector that represents the boundary conditions imposed by the pixels $above(u)$, $below(d)$, to the $left(l)$ and to the $right(r)$ of the current block.

$$\begin{aligned} c(0,0) &= l(0) + u(0) \\ c(0,B-1) &= r(0) + u(B-1) \\ c(B-1,0) &= l(B-1) + d(0) \\ c(B-1,B-1) &= r(B-1) + d(B-1) \end{aligned}$$

Other sophisticated error concealment techniques have been proposed in the literature, especially for video. For example, in [35], the authors propose the use of a Markov Random Field image model and optimally interpolate the missing pixels. The emphasis of our scheme, however, is on simplicity and on maximizing interactivity, rather than precision, for which we find empirically that our simple averaging strategy seems to work well.

4.4 Other Formats

We have described a simple framing strategy and further refinement using error concealment scheme for JPEG over ITP. The same techniques also extend to progressive JPEG images. In progressive JPEG, the quantized DCT coefficients corresponding to each block are divided into a series of scans. These scans may either represent different frequencies (low to high), or different bit-planes of the quantized coefficients (most significant to least significant bits). A coarse representation of the image is rendered with the receipt of the first scan, which is successively refined as subsequent scans arrive. Each scan can be segmented into restart intervals, which results in the ability to process and render out-of-order within a scan, leading to quicker response times and interactivity. Error-concealment can be carried out in a multi-resolution manner by performing concealment within one scan at a time.

Similar techniques are also possible for transmission of JPEG2000, which is a recent proposal for wavelet-based image coding scheme that results in higher compression ratios and better fidelity. The standard supports several features such as layered coding and "region of interest" (ROI) coding. Designing transport support for ROI coding requires customized scheduling of retransmission requests at the receiver, which is provided by ITP.

5 Performance Evaluation

In this section, we evaluate our implementation of ITP under a variety of network loss rates. Our implementation of ITP performs out-of-order data delivery at the receiver and uses the averaging method to interpolate missing packets at the receiver. We have customized ITP for JPEG transport where the images contain restart intervals. We have not implemented nor evaluated other formats We first discuss the performance metrics we use and present the results of our evaluation.

5.1 Peak Signal-to-Noise Ratio (PSNR)

Image quality is often measured using a metric known as the PSNR, defined as follows. Consider an image whose pixel values are denoted by $x(i,j)$ and a compressed version of the same image whose pixel values are $\hat{x}(i,j)$. The PSNR quality of the compressed image (in dB) is:

$$\text{PSNR} = 10 \times \log_{10} \frac{255^2}{E||x(i,j) - \hat{x}(i,j)||^2} \tag{4}$$

In our experiments, we use PSNR with respect to the transmitted image as the metric to measure the quality of the image at the receiver. Note that PSNR is inversely proportional to the mean-square distortion between the images, which is given by the expression in the denominator of Equation 4. When the two images being compared are identical, e.g., at the end of the transfer when all blocks from the transmitted image have been received, the mean-square distortion is 0 and the PSNR becomes ∞. We recognize that PSNR does not always accurately model perceptual quality, but use it because it is a commonly used metric in the signal processing literature.

5.2 Experimental Results

We measure the evolution of instantaneous PSNR as the JPEG image download progresses. When JPEG-ITP receives a complete restart interval from ITP, it is passed to the decoder. The decoder output is processed to fill in missing intervals using the error concealment step explained earlier and the image is updated. We measure PSNR with respect to the original JPEG image transmitted under three scenarios: (i) when TCP-like in-order delivery is enforced, (ii) when out-of-order delivery is allowed, and (iii) when error concealment is performed on the mis-ordered data units.

Figure 9 shows the results of this experiment under a variety of loss rates. We use a simple Bernoulli loss model where each packet is dropped at the receiver with an independent probability given by the average loss rate.

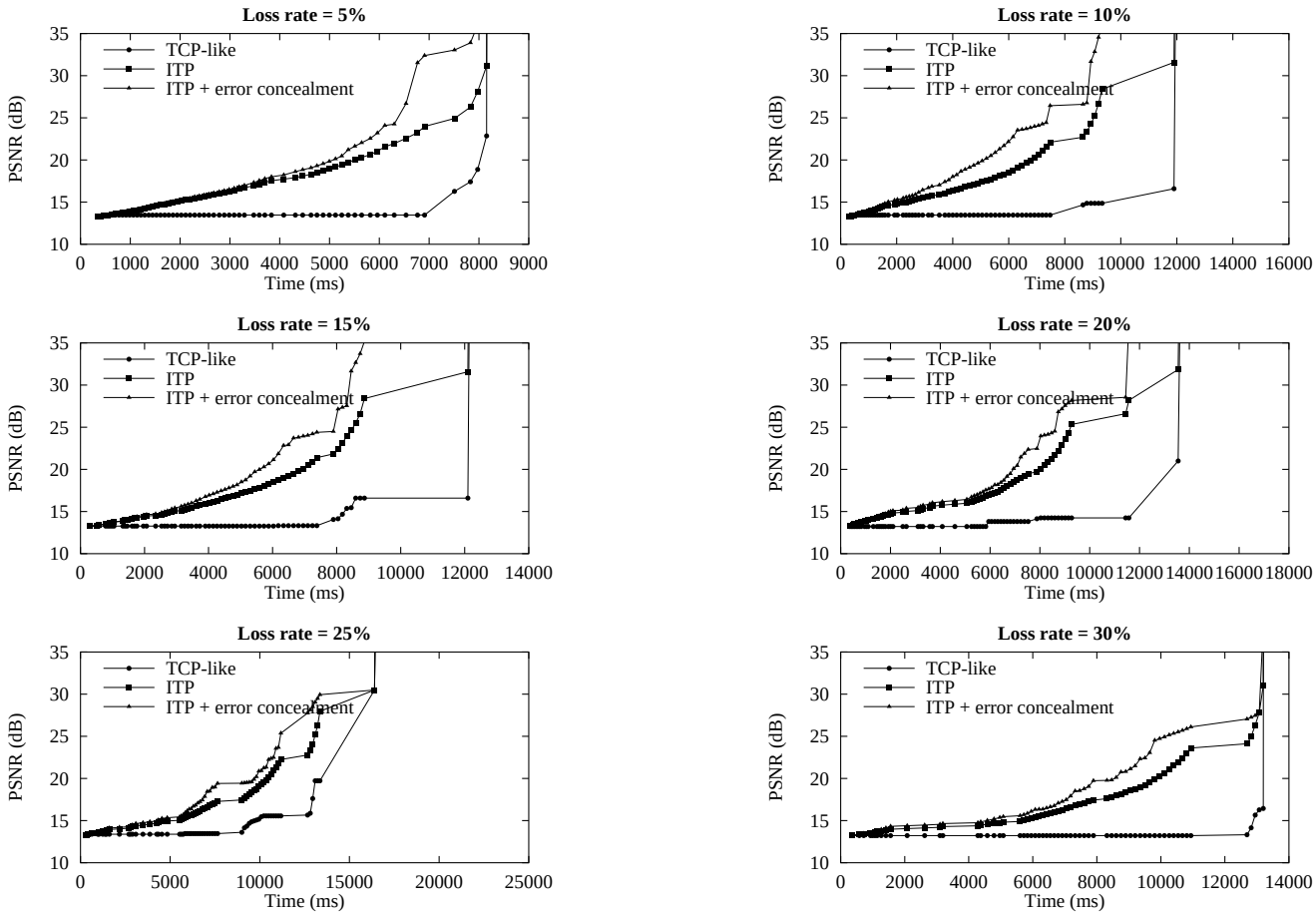

Figure 9. PSNR vs. Time for ITP and a TCP-like transport that enforces in-order delivery. The quality of the image (as measured by PSNR) is identical in all three scenarios at the start and at the end of the transfer. However, the sample paths differ — the best performance is seen with ITP optimized with error concealment, while TCP shows the poorest performance. ITP shows a steady improvement in quality, and is therefore perceptually superior for interactive applications such as the Web.

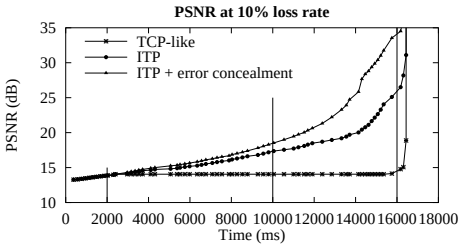

Figure 11. PSNR corresponding to the snapshots shown in Figure 10. Starting at almost identical image snapshots at $2s$, the ITP image (with and without error concealment) progress steadily in quality, while the TCP-delivered image only catches up close to completion time.

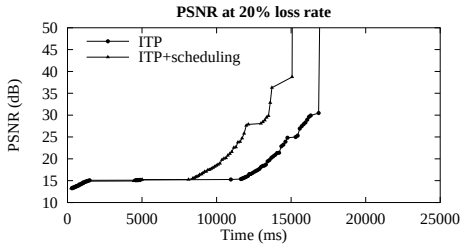

Figure 12. When receiver request scheduling takes into consideration those "stripes" that cannot be interpolated, the quality of the rendered image can be improved by 5–15 dB.

We find that across a range of loss rates between 5% and 30%, TCP-like delivery causes the quality of the rendered image to remain low for extended intervals of time. In comparison, ITP with out-of-order delivery shows a smoother evolution of PSNR during the transfer. In addition, the PSNR of the ITP-delivered image is superior to that delivered by TCP while the transfer is in progress, becoming identical only at the end of the transfer, as expected. This smooth evolution of quality makes ITP better suited for interactive image downloads. When error concealment is applied as an added optimization on the partial image, we find that the benefits are between 2–8 dB. In combination, the two techniques outperform TCP by 10–15 dB.

Figure 10 shows the progression of displayed images for the three different scenarios and Figure 11 shows the corresponding PSNR values. Starting with almost identical image snapshots at $2s$, the ITP-delivered images (with and without error concealment) show steady improvement in quality relative to the TCP-delivered snapshot. At $10s$, the ITP image is 3.3 dB and a further improvement of 1.3 dB is achieved through interpolation on the partial image. As we can see from the image, the benefits of interpolation are greater when more of the image is available, which further strengthens the case for out-of-order delivery in ITP. The ITP images continue to improve and at $12s$, they are 12 dB (without error concealment) and 20 dB (with error concealment) better than the corresponding TCP-delivered images. We also conduct a transfer across a 1.5 Mbps link to study the effect of receiver scheduling. Here, the receiver prioritizes requests for data items that cannot be concealed using interpolation.

In summary, we find that the rate of increase in PSNR with time is significantly higher for ITP compared to TCP-like delivery.

6 Related work

The so-called CATOCS debate on ordering semantics in the context of multicast protocols drew much attention a few years ago [6, 5, 7]. Cheriton and Skeen argued that ordering semantics are better handled by the application and that enforcing an arbitrarily chosen ordering rule results in performance problems [6]. In our work, we reinforce this approach to protocol design and refrain from imposing a particular ordering semantics across all applications.

RDP [41, 25] is a reliable datagram protocol intended for efficient bulk transfer of data for remote debugging-style applications. RDP does not enforce ordered delivery unless specified by the application. It implements sender-driven reliability and does not support receiver-tailored nor application-controlled reliability. NETBLT [8] is a receiver-based reliable transport protocol that uses in-order data delivery and performs rate-based congestion control.

There has been much recent work on Web data transport for in-order delivery, most of which address the problems posed to congestion control by short transaction sizes and concurrent streams. Persistent-connection HTTP [24], part of HTTP/1.1 [11], attempts to solve this using a single TCP connection, but this causes an undesirable coupling between logically different streams because it serializes concurrent data delivery. The MEMUX protocol (derived from Web MUX [13] proposes to deliver multiplexed bidirectional reliable ordered message streams over a bidirectional reliable ordered byte stream protocol such as TCP [44]. We note that the problem of shared congestion control disappears when congestion state is shared across TCP connections [1, 23, 39] or more generally, across all protocols using the CM.

The WebTP protocol argues that TCP is inappropriate for Web data and aims to replace HTTP and TCP with a single customizable receiver-driven transport protocol [16]. WebTP handles only client-server transactions and not other forms of interactive Web transactions such as "push" applications. It is not a true transport layer (like TCP) that can be used by different session (or application) protocols like HTTP or FTP, since it integrates the session and transport functionality together. In addition, WebTP advocates maintaining the congestion window at the receiver transport layer, which makes it hard to share with other transport protocols and applications.

In contrast, our work on image transport is motivated by the philosophy that one transport/session protocol does not fit all applications, and that the only function that *all* transport protocols *must* perform is congestion management. The Congestion Manager (CM) extracts this commonality

Figure 10. Snapshots of the displayed image with a TCP-like transport (first row), with ITP (second row), and with ITP enhanced with error concealment (last row) at 10% loss rate. The entire transfer of the 184 KB image takes $16.57s$ to complete.

into a trusted kernel module [2], permitting great heterogeneity in transport and application protocols customized to different data types (e.g., it is appropriate to continue using TCP for applications that need reliable in-order delivery and RTP/RTCP over UDP for real-time streams, etc.). The CM API allows these protocols to share bandwidth, learn from each other about network conditions, and dynamically partition available bandwidth amongst concurrent flows according to user preferences.

We now briefly present an overview of transport protocols tuned for spatially structured data types such as images. While much work has been done on video transmission, image transport has received little attention despite constituting a large fraction of Internet traffic. Turner and Peterson describe an end-to-end scheme for image encoding, compression, and transmission, tuned especially for links with large delay [40]. As a result, they develop a retransmission-free strategy based on forward error correction. Han and Messerschmitt propose a progressively reliable transport protocol (PRTP) for joint source-channel coding over a noisy, bandwidth constrained channel. This protocol delivers multiple versions of a packet with statistically increasing reliability and provides reliable, ordered delivery of images over bursty wireless channels [17]. The Flexible Image Transport System (FITS) is a standard format endorsed by the International Astronomical Union for the storage and transport of astronomical data [12]. It specifies various file header formats, but not a protocol for transmission over a loss-prone network.

The Fast and Lossy Internet Image Transmission protocol (FLIIT) [10] improves the perceived delay of a download by eliminating retransmissions. Instead, the FLIIT sender strategically shields "important" portions of the image data, for example, by applying FEC to the high order bits of the DC channels of the image. FLIIT assumes a bit budget and allocates this between the original image data and the amount of redundancy based on the observed loss rate in the channel. Our work in this dissertation also aims to improve interactivity. However, rather than design new compression schemes for image transmission, we focus on the transport protocol and application interface issues such that many different image formats can be supported.

Heybey [18] considers the problem of video coding and develops an application-level framing architecture for it. However, much emphasis is placed on developing framing strategies that translate into an optimized hardware implementation. In our work, we focus on the protocol aspects and show how a generic protocol may be used effectively when customized for specific image formats.

Richards and others [34] have also considered using ALF to build high performance transport protocols. However, they attempt to extend existing TCP implementations to achieve this and present their evaluation of the overheads involved in this approach.

Finally, we observe that several highly sophisticated error concealment techniques have been proposed in the literature, especially for video. For example, in [35], the authors propose the use of a Markov Random Field image model and optimally interpolate the missing pixels. The essence of our scheme, however, is on simplicity and maximizing interactivity (rather than precision), for which we find empirically that our simple averaging strategy seems to work well.

7 Conclusion

In this paper, we observe that the reliable, in-order byte stream abstraction provided by TCP is overly restrictive for richer data types such as image data. Several image encodings such as sequential and progressive JPEG and JPEG 2000 are designed to handle sub-image level granularities and decode partially received image data. In order to improve perceptual quality of the image during a download, we propose a novel Image Transport Protocol (ITP). ITP uses an application data unit (ADU) as the unit of processing and delivery to the application by exposing application framing boundaries to the transport protocol. This enables the receiver to process ADUs out of order. ITP can be used as a transport protocol for HTTP and is designed to be independent of the higher-layer application or session protocol. ITP relies on the Congestion Manager (CM) to perform safe and stable congestion control, making it a viable transport protocol for use on the Internet today.

We have shown how ITP is customized for specific image formats, such as JPEG. Out of order processing facilitates effective error concealment at the receiver that further improve the download quality of an image. We have implemented ITP as a user-level library that invokes the CM API for congestion control. We have also presented a performance evaluation demonstrating the benefits of ITP and error concealment over the traditional TCP approach, as measured by the peak signal-to-noise ratio (PSNR) of the received image.

In summary, ITP is a general purpose selectively reliable transport protocol that can be applied to diverse data types. Our design and implementation provide a generic substrate for congestion-controlled transports that can be tailored for specific data types.

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