             A Comment on Packet Video Remote Conferencing and the
                           Transport/Network Layers

Status of this Memo

   This memo provides information for the Internet community. It does
   not specify an Internet standard. Distribution of this memo is
   unlimited.

Abstract

   The new generation of multimedia applications demands new features
   and new mechanisms for proper performance. ATM technology has moved
   from concept to reality, delivering very high bandwidths and new
   capabilities to the data link layer user. In an effort to anticipate
   the high bandwidth-delay data link layer, Delta-t [Delta-t], NETBLT
   [RFC 988], and VMTP [RFC 1045] were developed. The excellent
   insights and mechanisms pioneered by the creators of these
   experimental Internet protocols were used in the design of Xpress
   Transfer Protocol (XTP) [XTP92] with the goal of eventually
   delivering ATM bandwidths to a user process. This RFC is a vehicle
   to inform the Internet community about XTP as it benefits from past
   Internet activity and targets general-purpose applications and
   multimedia applications with the emerging ATM networks in mind.

1. Introduction

   Networking is no longer synonymous with analog telephony. High-
   performance lower-layer networks have made possible exciting new
   applications: collaboratory environments, distributed client/server
   computing, remote conferencing, teleclassrooms, and distributed
   life-sciences imaging. These applications normally demand a great
   deal of bandwidth and often create operating system bottlenecks.
   Enabling these new multimedia applications entails delivering
   bandwidth to the applications, not just having bandwidth available on
   the network. This statement may appear obvious, but often solutions
   at the transport layer are satisfied by having bandwidth at that
   layer without sufficient sensitivity to higher-layer access to the
   bandwidth. The unavailability of bandwidth at upper layers is
   becoming the real issue as the networks are becoming a high-
   performance virtual backplane without concomitant high-performance
   control schemes. It appears that new services are needed that
   require communication with all layers. The ATM architecture calls

   for such an integrated control scheme.

2. Remote Conferencing

   The challenges of remote conferencing is an application whose
   challenges may be met at the data link layer by the emerging
   broadband networks. If so, important medical applications such as
   medical imaging for diagnosis and treatment planning would be
   possible [CHIM92]. Remote conferencing would permit imaging
   applications for life sciences through the use of national resource
   centers. Collaboratory conferences in molecular modeling, design
   efforts, and visualization of data in numerous disciplines could
   become possible.

   At the Second Packet Video Workshop, held December, 1992, at MCNC in
   the Research Triangle Park, North Carolina, a recurrent theme was the
   use of multimedia in remote conferencing. Its applications included
   the use of interactive, synchronized voice and video transmission,
   multicast transmission, data transfer, graphics transmission,
   noninteractive video and audio transmission, and data base query
   within a virtually shared workspace. A few participants doubted the
   ability of current computer networks to handle these multimedia
   applications and preferred only connection-oriented, circuit-switched
   services. Most participants, however, looked forward to using an
   integrated network approach.

2.1. Remote Conferencing Functions and Requirements

   Remote conferencing as seen at the workshop requires a set of
   functions. It must provide session scheduling that deals with
   initiating a session, joining in-progress sessions, leaving a session
   without tearing it down if there are multiple participants, and
   terminating a session.

   The remote-conferencing session needs a control subsystem that is
   either tightly controlled for an n-to-n connection for two to 15
   participants, or loosely controlled for a 1-to-n connection for any
   number of participants. The Multipeer-Multicast Consortium is
   working on defining the control requirements and the mechanisms for
   control. At the Packet Video Workshop, one participant presented a
   conference control protocol (CCP) shown in Figure 1 [CCP92]. In this
   architecture the CCP controls the Network Voice Protocol (NVP)
   [RFC741] and the Packet Video Protocol (PVP) [PVP81] over the
   experimental Internet Stream Protocol, Version 2 (ST-II) [RFC1190]
   rather than IP.

   Latency and intramedia synchronization and intermedia synchronization
   (lip-sync) are critical for the interactive voice and video streams

of remote conferencing. Client/server applications including data
base operations are equally important. The ability to display
noninteractive video and high-resolution graphics is necessary.

As with most applications, security will eventually be an issue. At
the very least, there must be a mechanism to determine who can find
out what about conference and who can join a conference. This
determination will be part of an upper-layer protocol.

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    +--------------+ +--------+ +-----+ +------------+
    |Teleconference| |  File  | |Email| |   Domain   |
    |    (CCP)     | |Transfer| |     | |Name Service|
    +----+-------+-+ +-----+--+ +-+---+ +-----+------+
         |       |        |      |__  __|           |
         |       |        |        ||              |
   +-----+--+ +--+-----+    +-++-+          +----+---+
   |Network | | Packet |    | T |          |   U    |
   | Voice  | | Video  |    | C |          |   D    |
   |Protocol| |Protocol|    | P |          |   P    |
   +---+----+ +--+-----+    +-+--+          +--+-----+
       |__    __|            |__         __|
          |   |               |           |
       +-+---+--+          +-+-------+-+
       | Stream |          |    I      |
       |Protocol|          |    P      |
       +---+----+          +---+-------+
           |                   |
    +-----+--------------------+----+
    |IEEE_802.X,FDDI,DARTnet,ATOMIC...|
    +--------------------------------+
```
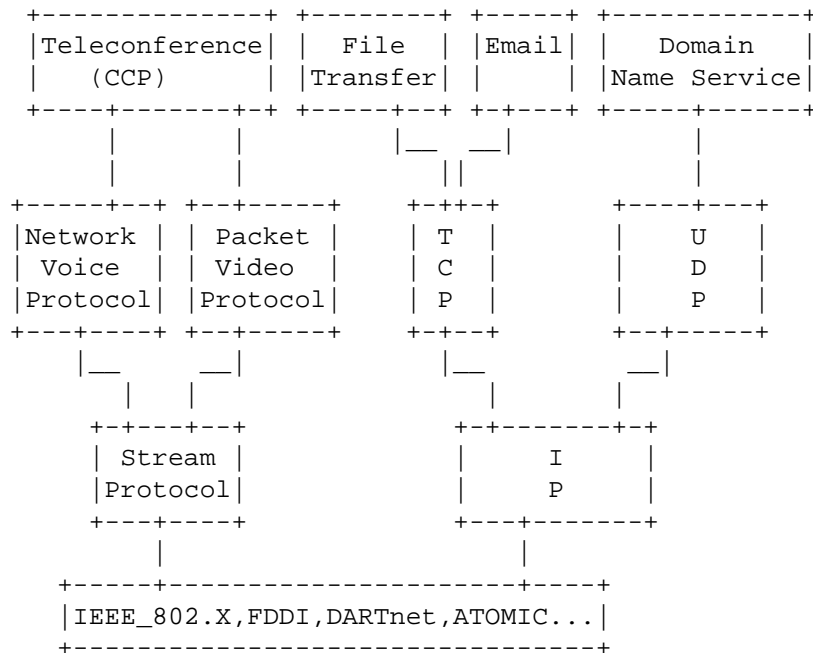
            Figure 1: The Connection Control Protocol Architecture

The solutions must range in geography from single machines through
LAN, CAN, MAN, WAN conferences, as well as in data content from PCs
to high-end workstations. Implicit in the scaling is the notion of
product and application interoperability.

Remote conferencing applications should take advantage of future
network enhancements, as well as the functions provided by ATM, FDDI,
and SMDS. Doing so should provide function versus resource trade-
offs such as cost versus error control and cost versus rate control.
As a result, the transport layer should be able to provide the
services offered by the data link layer.

Most of the presentation on remote conferencing indicated a need for
some degree of multicast functionality, ranging from the 1-to-n, with
conference membership completely known, to conferences for which

   existence of a group is known, but exact membership is not.

   In remote conferencing, it is preferable to use one network for all
   information traffic. This network should have an offered quality of
   service (QOS) criteria to accommodate tradeoff metrics, which would
   include guaranteed throughput, connection reliability, high call-
   completion rate, few dropped calls, tolerable error rate, varying
   degrees of compression on the video and audio streams, and tolerable
   motion artifacts, flow control, and latency. The QOS should be an
   input to the transport layer from the application or transport
   service user.

   The compression/coding function should provide time-stamping and
   packetizing information, as well as real-time echo cancellation.
   These functions are usually at the presentation and session layer of
   the Open System Interconnection (OSI) model or the at the application
   in some Internet models, but not the transport layer.

3. Potential Solutions

   RFC 1193 deals with the requirements of real-time communications,
   which include some of the same capabilities [RFC1193]. But the
   requirements articulated create the necessity for new
   transport/network protocols. The new protocols under development by
   the Audio Visual Transport [SCHU92] (RTC, RTCP), and other protocols
   in this area (ST-II) suggest an architecture like that shown in
   Figure 2.

   These approaches may work. However, they encourage a discipline that
   creates a protocol for each new class of application. Another
   approach might be to unify the protocols. It is felt that this is
   one of the main goals of XTP (see Figure 3).

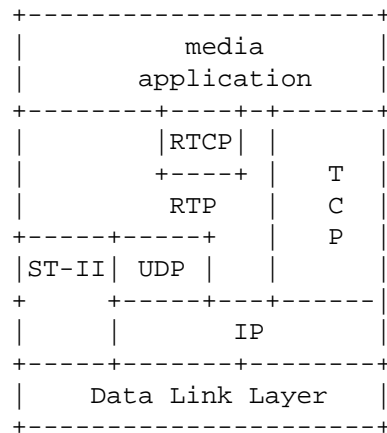   Other design considerations of XTP include the following:

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              +----------------------+
              |        media         |
              |     application      |
              +--------+----+-+------+
              |        |RTCP| |      |
              |        +----+ |  T   |
              |        RTP   |  C   |
              +-----+-----+  |  P   |
              |ST-II| UDP |  |      |
              +     +-----+--+------|
              |     |       IP      |
              +-----+-------+--------+
              |   Data Link Layer   |
              +----------------------+
```

              Figure 2: One emerging multimedia architecture


```
   +--------------+  +--------+ +-----+ +------------++-----------+
   |Teleconference|  | File   | |Email| |  Domain    ||  media    |
   |              |  |Transfer| |     | |Name Service||application|
   +------+-------+  +----+---+ +--+--+ +-----+------++-----+-----+
          |             |         |        |           |
          +--------------+--------+---------+-------------+
                         |
               +-------+--------+
               |Unified Protocol|
               +----------------+
               |Data Link Layer |
               +----------------+
```
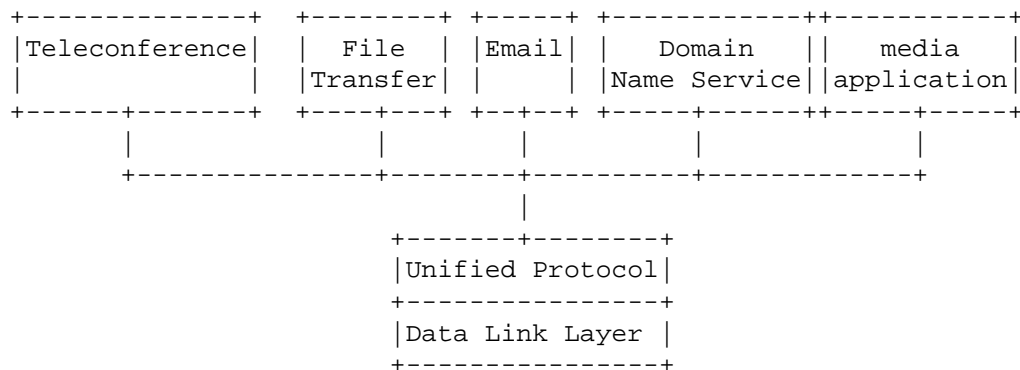
          Figure 3: Another integrated multimedia architecture

   (1) Developing a protocol based on the work and experience of
        the protocol groups such as IETF, which produced NETBLT,
        VMTP, TCP, IP, and UDP.

   (2) Incorporating lessons from Delta-t.

   (3) Observing the design paradigm shift of ATM, SONET, and VMTP
        in the header, trailer, and information segment construction.

   (4) Targeting the real-time requirements articulated by the
        Department of Defense SAFENET committee and the French
        Ministry of Defense military real-time specification [GAM-T-103].

   Mechanisms in XTP allow an application to approach the performance
   desired. A session-scheduling mechanism for joining and leaving a

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multicast conference exists.  The XTP mechanism for multicast
satisfies the loosely controlled session requirements.  The tightly
controlled session would require the use of multiple XTP
associations.

The priority mechanism that uses the 32-bit SORT field permits an
application to prioritize data.  Because XTP is a transport layer,
this priority mechanism follows through every node tranversed.  There
is also an out-of-band delivery mechanism.  However, XTP does not
offer latency control by itself; it just provides a priority
mechanism.

The selective acknowledgement, fast negative acknowledgement, and
selective retransmission permit an application to choose an
appropriate level of error control.  The combination of the priority
mechanism and these error-control mechanisms is likely to approach
the latency and synchronization requirements of remote conferencing.

Noninteractive audio and video, as well as graphics presentation, can
be accommodated in many ways by the application.  It is important
that the transport layer provides adequate mechanisms to deliver the
appropriate data streams in a manner compatible with the
applications.  These applications can probably be accomplished by
means of extant protocols, as well as XTP.

The scalability of the solution will be a function of the standards
used.  At the Packet Video Workshop, some of the applications
sacrificed computer network standards in favor of desired
performance.  This approach usually impedes scalability.  From the
explanation of the applications taking this approach, it appeared
that using XTP would have provided an adequate transport service for
the applications.

XTP was designed to provide mechanisms to accommodate application
requirements, that is, the ability to respond to QOS requests.  For
example, guaranteed throughput may be accomplished by using XTP's
rate and burst control together with flow control or no flow control.
Tolerable error rate can be accomplished with partially error
controlled connections (PECC), a service which can be placed in the
application or just above the transport layer [PECC92].  Motion
artifacts and varying degrees of compression should be done above the
transport layer in coordination with the transport layer or possibly
in a network management function.

3.1.  Synthesize the Hardware Fabrication Process into the Design

To produce an affordable solution, the hardware fabrication process
should be a design consideration.  Technologies are evolving too

rapidly to assume that a generic protocol design will anticipate all
fabrication advances, but this fact should not impede use of the
features of advanced hardware fabrication processes.

System interface problems and VLSI techniques should be considered in
the specification of the protocol.  An examination of the ATM and
SONET standards appears to support this philosophy.  Similarly,
NETBLT and VMTP design efforts seem to support this approach.  XTP
does use it.

It is very helpful to break down the protocol into parallel-state
machines for execution on more inexpensive hardware.  This procedure
reduces the context switching and interrupt handling requirements of
the hardware, thereby decreasing production costs while producing a
scalable protocol machine.

4.  Multimedia Applications over XTP

In parallel with the IETF efforts to enable multimedia applications
such as remote conferencing, the XTP forum members have been
experimenting with major elements of these applications.


(1)  At the University of Virginia, more than 100 simulated voice
     channels were run on an FDDI network [UVAVOICE92].  The
     purpose of this experiment was to test the limits of FDDI
     and a software version of XTP in a simulated interactive
     voice environment.  Multicasted, noninteractive video has
     been supported there for several years.

     UVa also has a video-mail demo over XTP/FDDI that uses
     Fluent multimedia interface and standard JPEG compression.
     This PC-based demo delivers full frame, full color, 30
     frames/sec video from any network disk to a remote VGA
     screen.  It is important that users could not discern any
     difference  in  playback  between  a local disk and a remote
     disk.  An Xpress File System (XFS) is used on server and
     client systems.

(2)  The Technical University of Berlin, Germany, reports that
     the coordination, implementation, and operation of
     multimedia services (CIO) of the R&D in Advanced
     Communications Technologies in Europe (RACE) is using XTP as
     a starting point for design [XTPRACE].

(3)  At the Naval Command, Control, and Ocean Surveillance Center
     Research, Development, Test and Evaluation Division NRaD
     (formerly the Naval Ocean Systems Command (NOSC)), voice is

multicasted over XTP/FDDI.  A simple multicast is
distributed to a group with a latency of around 25 ms, where
the latency represents delay from the voice signal from the
microphone to the audio signal to the speaker.  This group
is currently doing research on an n-party multicast of voice
(telephone conference-call paradigm [n x n multicast]).

(4)  Commercially, Starlight Networks Inc., migrated a subset of
XTP into the transport layer of its video application
server.  By using XTP rate control, full-motion, full-screen
compressed video is delivered at a constant 1.2 Mbps, over
switched-hub Ethernet to viewstations.  This network
delivers at least 10 simultaneous video streams.

Therefore, XTP has been used in applications that were previously
placed over IP or even a data link layer.

5.  Policy versus Mechanism

Separating protocol policies and mechanisms [XTPbk] permits adoption
of new policies without altering offered mechanisms.  An excellent
example is UVa's Partially Error Controlled Connections (PECC).  This
control system maximizes error control in such a way that receiving
FIFOs are never starved i.e., the application, driver, or operating
system buffer control, and not the transport layer becomes the
bottleneck.

Because XTP is mechanism-rich and policy-tolerant, this very dynamic
error control policy mechanism is possible.  Separating policy and
mechanism permits an error-control or flow-control policy to adapt to
the data link layer conditions without shutting down a connection and
rebuilding (or multiplexing) a new one on a different protocol stack.
This may also provide an easier way for a network management
subsystem to maintain a desired QOS.

6.  Summary

Remote conferencing presents new opportunities for research,
business, and administration.  Although some are proposing that only
classical circuit-switched mechanisms be used, most network engineers
are searching for ways to use the new features of FDDI, SMDS, and ATM
in multimedia applications such as remote conferencing.  Some new
applications have been placed directly on a data link layer.  New
Transport/Network layer combinations have been proposed and are being
tested.  It is believed that consideration should be given to XTP as
a possible solution because its forum membership includes commercial,
government, and research institutions, some of which have implemented
various applications that contribute to an overall remote-

conferencing application.

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Security Considerations

   Security issues are discussed in section 2.1.

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