Networking Technology for Multimedia - DARPA, RACE and PACT Experiences. RN/92/XX

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ABSTRACT

The Department of Computer Science, University College London, is involved in a number of multimedia projects, ranging from pilot to service, from Laboratory scale to International, from transmission and coding technology to shared workspaces.[1] This paper describes some of the experience gained in these projects.

1. Introduction

The Department of Computer Science, University College London, is involved in a number of multimedia projects, ranging from pilot to service, from Laboratory scale to International, from transmission and coding technology to shared workspaces.

- SERC (CDS) PACT - Video Coding and Traffic Control

  This is a small project concerned with the interaction between video and audio coding schemes and network traffic control mechanisms. We have devised an abstraction (non-layered) for capture/compression/transmission/reception/decompression/display which we have used to compare different coding schemes and their adaptability. We have also examined the interaction between multi-party conferences and hierarchical video coding schemes.

- DARPA

  Under a continuing DARPA funded project we have been looking at two different structures for video-conferences:

  - BBN/ISI Conference Room System

    BBN and ISI have devised a system for connecting together conference rooms - typically each room has a number of people, several cameras, monitors or projection devices, and mikes/speakers. Cameras and video displays are connected to a local video switch, which has a CODEC attached. The CODEC is plugged into a packetiser and to an Internet Router. The router provides real time support (at least jitter control and priority) for video packets via flow specifications using the ST protocol.[2]

    Both floor controlled (with only the speaker visible) and "Hollywood Square" style conferencing have been used.

  - DARTNET Style Workstation Conference

1. This work was supported in part by DARPA under Contract Number N00014-86-0092, and by the RACE program under the CAR and PREPARE projects, and by the SERC CDS ctte under the PACT project. Any views expressed in this paper are those of the author alone.

2. In the case of the London site, the analogue video network actually covers most colleges throughout the capital, and the same cable carries the packets between routers as well in a hybrid broadband analog/digital network.

A more loosely structured conferencing system uses (barely modified) Internet Datagrams to carry audio and video on a best effort - usually at constant bit rate (CBR). A lot of design work on transport protocols for end to end description of the media coding and receiver re-sequencing/recovery of timing information. Very dynamic large audio conferences have been built (200 participants world wide) with smaller number of low quality video participation. Typically, such conferences have been used for *dissemination* of a large meeting.

- RACE CAR and PREPARE - Shared Applications - Collaborative Computing

  Under the RACE program, UCL in the CAR project have built a distributed collaborative design system. The main area of interest was in the ability for Car designers from major design houses and automobile manufacturers to be able to work together remotely using tools that they were familiar with. In the CAR Project, either analog video and audio or low quality H.261 compressed video over ISDN circuits was used along with packet switching to support shared windows and shared applications.

2. Audio and Video Coding

Audio sources are typically Erlang distributed - the simplest model (two markov state) is talk spurts and silences. Audio is captured in a or mu law PCM - it can then be compressed by a variety of techniques (e.g. differential coding, adaptive differential or linear excited predictive coding schemes). Most of these still end up offering CBR traffic or no traffic. The load can be as low as 4kbps (e.g. using Linear Predicitive Coding) or as high as 1.4 Mbps (e.g. CD quality stereo).

Video coding techniques are more varied, and fall into two main categories:

1. Constant Bit Rate
2. Constant Quality

It is easier to engineer traffic control mechanisms to support Constant Bit Rate since this can simply be decided on the basis of call admission and static bandwidth availability. If the CBR traffic is serviced with sufficient frequency, then a playout buffer (this works for audio too) can accommodate any variation in the net within reason. Occasional real loss can be sustained by interpolation of audio or leaving the previous video in place.

Variable bit rate video is much harder to deal with in the network (although it is a far more natural model of the source, if it is a natural scene).[3]

Experience with the Internet shows that rapidly varying traffic is more disruptive than simply high load traffic. Certainly during a video+audio broadcast, camera shake/zoom/pan would cause traffic variations that caused the audio to break up. Note that other traffic is predominantly TCP (apart from DNS and possibly ill-advised Wide Area NFS traffic) - TCP will adapt to the bandwidth available in a controlled manner, so is not badly affected by CBR traffic.

Video codes currently in use include X Image based, run length coded, differenced blocks of bitmaps, H.261 DCT coded, and DVI coded. We await the availability of MPEG chips, as there is certainly little chance of implementing an MPEG coder in software for a few years.

3. Multimedia Protocols

Having said that we were using IP au naturel, in fact we also had a number of mechanisms for ensuring some hope of success.

3. Note that we have used video to disseminate slides from many meetings - it is very useful since it doesn't require preparation of slide materials in advance (or in a form that the speaker/author may not be familiar with).

1. ST

   The DARPA TWBNet conferencing system used the ST protocol. ST provides flow definition as well as multicast addressing - it is a connection oriented replacement for IP. It uses subnet technology to achieve flow guarantees rather than providing mechanisms itself.[1]

2. IP over X.25

   The JANET IP Service is run above the Janet II 2 Mbps X.25 service. It is an unusual X.25 since internally between switches it is implemented without hop by hop flow control. The IP routers sit on the outside of a collection of X.25 switches, and are given a share of the network determined by the clock rate of the local interfaces to the switches, and by the packet size/windows of the VCs between routers.

3. Path (over-)engineering

   For workstation conferencing, we have used IP *un-adulterated*. However, very careful path engineering of the multicast IP topology is required.

4. Multicast

A Multicast Datagram Network Service in Wide Area Networks is a relatively new technique based on forming host groups and enhancing routers to form special purpose forwarding information bases to these groups. This is mainly so that the number of copies of packets sent to replicated services may be reduced to near optimum in the same manner as on networks with a physical broadcast technology (e.g. Satellite and Ethernets).[2,3,4]

Multi-media conferencing is an application that often involves more than two end systems, and could take advantage of a multicast network service. However, multimedia systems often employ application level distribution. For instance, the voice portion of a conference is often sent by each participant to a central mixing service, which then sends the mixed portion to each member. To be optimally placed for each client, the mixing server may well be pessimally placed to use multicast to redistribute the mixed voice, compared with a forest of multicast trees rooted at each sender, or rooted at each destination group.[4] With the video portion of a conference, it is often quad multiplexed by CODECs, in a similar manner to audio mixing, or else bandwidth requirements preclude anyone but the floor holder of a conference being visible.[5,6,7,8]

Clients of Replicated Databases distribute the same updates and retrieve the same records from more than one copy of the service and could also make use of a multicast network service. However, many transaction systems involve 2 or 3 phase commit protocols with each server in the replicated system for writing (although for reading, a majority read may suffice - however we need to know what the majority is).[5]

4. Optimally placed to receive and return from and to each client requires that we minimise the distances, and place the mixer in the centre of the clients. Now unless packets start going different paths when they are unicast to when they are multicast, the optimal place for unicast is also going to be optimal for multicast (unless someone can come up with a counter-example). Of course this may not be true if we are replacing the n-1-n with n-n distribution and let each destination mix as required.

5. Multicast can also be used for user/conference location. User location involves identifying the current physical location of a given user, and provide a invitation service as needed (or as Cosmos called it, User Locator) In this case, there are two modes that could make use of multicast:
   a) a multicast "rwho" that the conference service cache the result.
   We can also use this to find out the people we can conference
   with.
   b) do a multicast to find out about the user when needed.
   Stations that has multimedia conference capabilities can then join the multicast group. Alternatively, only users that wants to be found will join the multicast group.

   Conference finding: We can either use multicast to construct a conference finding service. Or we can use that as an optimisation between a real directory system and the application. All conference instance will be stored in the directory and local ones cache at the conference finding daemon. Conference finding request will always be sent out on a local multicast address. Local conference will then return using the cache result at the conference finding deamon. Non local conference then lookup the real directory using DAP.

In the CAR Experimental platform, UCL support shared applications which can be either collaboration *transparent* or *aware*. We have recently started to add digital audio and video to the CAR system, and have had to slightly modify our view of what constitutes an **application** in the context of multimedia conferencing, as well as the extent to which our existing single model of conference management and of replication suffices.

5. Conference Room Systems

Conference rooms are expensive. They are also highly effective since a number of components needed for long haul (digital) (packet-switched) conferencing are also expensive - for instance high quality CODECs, intelligent multi-protocol routers, analogue video switches and remote video and audio control equipment.

6. Workstation Conferencing Systems

Workstation conferencing is relatively cheap now: since Keith Lantz performed the famous experiment using a childs videocamera, it has now reached the point where domestic quality video can be captured digitally at effective frame rates (>16 fps) and displayed in windows or transmitted at low compression rates. CPU and BUs speeds are not high enough yet to get high compression ratios as well as high frame rates - INRIA's h.261 in s/w can do 64kbps 1 frame per second on a 30MIPS sparc or mips processor. However, there are products in the pipeline (pun intended) that will provide block level transforms, motion detection/coding, and maybe more. However it will be another year or so before we can equip a machine with domestic quality video **and** sufficiently fast compression to let it loose on todays (2Mpbs/T1) wide area networks.

At the other extreme, audio is now completely within our grasp. Indeed, a modest 10 Mips workstation can even support multiple audio channels, mixing and muxing them as need be.

7. Shared Applications

In the CAR system at UCL certain decisions were made on distribution versus centralisation of services.

A shared editor can be implemented in 3 ways:

i. The data can be shared (e.g. through NFS) and each user runs their own copy of the editor - an out of band system negotiates read/write access.

ii. the interface can be distributed (e.g. shared X), the editor and data are both central. The window system permits input or output through an out of band (floor control) system.

iii. the application can be distributed, and manage aspects of read/write in an application specific way.

Similar design space choices have to be made for audio and video in multi-media distributed systems - e.g. is video all sent to one site, mixed into a ("Hollywood squares") single picture, and then cast out to all receivers, or is the signal (required - possible chosen through interest and/or floor control), propagated to all receivers, and all (or as many as sensibly fit) displayed? Similarly, audio can be sent to a single mixer site, then redistributed, or it can be cast to everyone, and every receiver mixes.

Which solution is chosen could easily evolve as we learn more about the performance (and consistency requirements) of such systems - certainly in since these systems directly serve humans, consistency is less a requirement than for programs (at least in current programming languages).

More interestingly, the floor control and negotiation of who can do what to which objects and when, together with decision support in such systems is in itself a multi-media distributed application, and possibly the most stressful.

8. Subjective Experiences

Full conference room systems actually can go so far as to replace some meetings, especially where technically focussed. Desktop conferencing, particularly audio with good indication of speaker, can provide an exciting replacement for bulletin boards. When augmented with slow scan video of overheads, it can replace attendance at a workshop or confernce almost perfectly, when offset against travel time and expenditure.

9. Summary and Future Work

We have broadly outlined the experience gained on a variety of Wide Area Multimedia systems, from coding, transmission, management and conference facilities.

Future UCL projects capitalising on this work will be based on two areas:

- The UK SuperJANET (B-ISDN) network.
- CEC pan-european packet video conferencing system

Apart from the engineering of QoS guarantees within the network, we feel that the single least well-understood/designed component of conferencing systems is management. This includes:

i. Delegate location/invitation protocols

ii. Booking systems (both for facilities and transmission resources).

iii. Floor Control and mixing/quad-multiplexing services

iv. Directories of conferences (a la bulletin boards/Internet Relay Chat structure).

v. GDMO Specification of Media Resources in end systems to support optimal choice of transmission versus coding/format converters for truly heterogeneous conferencing systems.

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