AARNET PROJECT SUBMISSION (4)

AARNET PROJECT FUNDING APPLICATION (1991 - 1992)

PROJECT TITLE:

Audio-Visual Broadcasting of Academic Conferences over AARNet.

INSTITUTION:

University of Sydney. Basser Department of Computer Science.

CHIEF INVESTIGATOR:

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BRIEF DESCRIPTION OF THE PROJECT.

Information and communication are the most useful resources in the academic and research environment. It is therefore important to make sophisticated forms of information distribution and communication readily available and easily accessible. AARNet, as a computer communications network, has attributes that often make it the best means of information distribution and communication within the academic community. Information is transferred digitally, communication delays are short, efficient (selective) broadcasting is possible and private network ownership keeps transmission costs low.

These properties are already exploited by electronic mail and news distribution services. However there are other highly useful academic services that can be efficiently provided by AARNet. This project aims to implement, experimentally over AARNet, a specific service which is perceived to be useful to a wide community: real-time broadcasting of the sound and vision of academic conferences to AARNet users who are not capable of attending the conference for one reason or another.

It is reasonable to assume that educators and researchers attend conferences to:

(a) listen to the presented talks, and

(b) discuss issues with other attendees.

Specifically, we attend a conference to hear and discuss a few talks which directly relate to our work and/or fall in the areas of our interest. It is also a fact that we often cannot afford to attend a conference because we may have other commitments at the time, we are only interested in a small portion of the conference or we cannot justify the time and the cost.

It is believed that AARNet can provide a service to allow academics to listen, via their computer terminal, to selected talks at remote conferences.

Real-time two-way transfer of sound and vision over computer communications networks requires large communications bandwidths and very low communications latencies and so is not able to be provided on general-purpose networks which are optimized for batch and interactive query traffic [1]. The specific service considered in this project is a one-way broadcast service and hence does not demand stringent communication delay requirements. Furthermore, the video traffic need only be a series of still images (talk slides) which require updating at a conservative rate of about one every fifteen seconds. This reduces the required video bandwidth greatly. Also, compression techniques for voice and video have advanced greatly in the last decade, reducing the required throughput to a fraction of the bandwidths of slow network links.

The aim of the project is to implement a pilot system on a restricted set of AARNet sites. The implementation involves:

- The design of a highly efficient voice/sound compression algorithm.
- The design of a suitable video image compression algorithm.
- Implementing the real-time transmission of voice and video over AARNet.
- Investigation of methods for efficient selective broadcasting and means of tapping into the service.
- Evaluation of the trade-offs between the quality of the service and the required communications bandwidths.

Technical Objectives:

1. *Sound compression.*

Transmission of PCM encoded speech over ISDN [2] requires a bandwidth of 64kbps. Application of advanced compression techniques can reduce the required bandwidth to less that 8kpbs. Transmissions of fair synthetic quality have been made at 2.4kpbs. We intend to pursue a combination of techniques including auditory and speaker modelling, sub-band and delta encoding, encoding to a error tolerances and entropy based methods to attempt acceptable speech quality at even lower bandwidths. High compression ratios can also make feasible digital storage of the sound and vision, allowing batch retrieval of selected talks at a later time. Voice messaging is also possible.

Decompression of the sound and vision information should be computationally simple to permit reproduction at the destination without the need for signal processing hardware (although this is increasingly being provided as standard on workstations.) Arbitrary waveform sound output hardware is common on PC's and workstations and sound input is now standard on some workstations while it is a low-cost option on PC's.

2. *Video Compression*

It has been demonstrated that even 64kbits per second can be used for slow motion real-time video transmission. For our purpose, we need to transmit only one image every 15 seconds, so the actual transmission rate is substantially reduced. In fact, a good presenter is not likely to change the slide for at least a minute. Coupled with advanced compression techniques, the amount of bandwidth required for the video should be less that one kilobit per second. We aim to display the video within a window on the host computer.

3. *Methods of Connection and Efficient packet broadcasting.*

Users will tap into the broadcast via a request to an advertised address. The user must then be added to the broadcast set in such a way to minimize the total communications band-width used. Exact methods of implementation will have to be investigated. We must also set packet priorities to ensure that remote login traffic - where low communications latency is important - is not slowed down by the sound and vision traffic.

PROJECT PHASES.

Phase 1: Research and Preparation

- Investigate and evaluate the best voice and video compression techniques,
- Investigate mechanisms for connecting to the broadcast stream and,
- Setting up necessary equipment such as voice and video digitizers for input data.

Phase 2: Design

- Software requirements analysis,
- Software structure for the implementation,
- Design tests and test specifications,
- Technical design specification.

Phase 3: Implementation

- Coding

Phase 4: Test and Evaluation

- Perform testing based on the test plan of phase 2.
- Evaluation and Documentation.

RESOURCE REQUIREMENTS:

Human Resources:

i) Chief investigator: D. B. Hoang.

ii) Programmer with relevant background in UNIX, Networking and Signal Processing to be employed.

Capital Equipment:

i) A voice digitizer,

ii) A video frame-grabber (or buffer),

iii) An IBM PC compatible,

iv) A signal processing design package and,

v) The AARNet and access to it.

PROJECT FUNDING REQUESTED.

Most of the items mentioned in the Resource Requirements are already available in the department, some terminal equipment is expected to be available associated with a remote AARNet site.

The specific items required for funding are:

1. It is requested that a programmer be funded for the purpose of implementation, testing and evaluation. The amount requested is a half-time salary for one year: $14 750 (1991-1992).

2. The purchase of a signal processing card with sound I/O for the IBM-PC ($500).

3. Funds to cover costs in testing at a remote AARNet site and to present results at an Australian Conference ($1000).

TOTAL FUNDING REQUESTED: $16 250.

Signature of Chief Investigator

Comments by Head of Department:

I fully support this project – will send further comment by email.

Signature of Head of the Department

R. J. Kummerfeld

REFERENCES

[1] NG,M.J.T and HOANG,D.B, "Joint Optimization of Flow and Capacity Assignment in Packet Switched Communications Network", *IEEE Transaction on Communications,* Vol. COM-35, No.2, Feb. 1987, pp.202-210.

[2] HOANG,D.B and WINDLE,J.," Packet Voice in Fast Packet Switched ISDN", *Fast Packet Switching Workshop Proceedings*, Telecom Australia Research Laboratories, Melbourne, May 19-20, 1988 .