Aspects of the Requirements and Implementation of a Booking Service for Internet-based Videoconferencing

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A dissertation submitted in partial fulfillment of the requirements for the degree of Master of Science in Computer Science in the University of Wales

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Acknowledgements

My sincere gratitude go first to my supervisor, Dave Price, whose patience and guidance has seen me through to the end of this piece of work, which has unusually taken longer than initially scheduled.

At the onset, the thought of delving into the area of videoconferencing over IP, an area still in it's infancy with scanty information available, was quite daunting. However, with the encouragement of my course mates (M.Sc's of 2000/1) and other colleagues in various capacities at the Department of Computer Science, I got the strength to get started. To you all colleagues, I say THANK YOU!!

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Finally, to my family, for their patience and having to put up with my absence during the course of my M. Sc.

DECLARATIONS

This work has not previously been accepted in substance for any degree and is not being concurrently submitted in candidature for any degree.

Signed ……………………………… (**Nasan Natseri**)

Date …………………………………

This dissertation is being submitted in partial fulfillment of the requirements for the degree of Master of Science in Computer Science.

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This dissertation is the result of my own independent work/investigation, except where otherwise stated. Other sources are acknowledged by explicit references to the bibliography. A bibliography is appended.

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Abstract

Videoconferencing over ISDN offers benefits such as available bandwidth, video frame rate and resolution, and audio quality, as well as speed of response when sharing applications, that are stable.

However, due to limited bandwidth as a resource, many institutions will have limited access to ISDN. At bandwidths in the lower range of 128 Kbps, the picture is delayed and there is severe lack of synchronisation between audio and video. Additionally, the equipment required under ISDN videoconferencing is expensive. Special rooms, which may be used for other purposes and each have different booking procedures, have been designed for this purpose, thus necessitating the users to move towards the technology. These resource restrictions have reduced the wider accessibility of this facility and meant that scheduling and booking of the different components of a video network are essential.

The core service, which allows users to connect to the MCU via the Public Services Telephone Network (PSTN), is free of charge to institutions. However, ISDN call charges are payable in connecting to the service. Even with the lowering of ISDN charges over the last couple of years, it is still not a cheaper transmission mode.

On the other hand, the usage of IP based videoconferencing is growing rapidly and has significant advantages such as low installation costs with elimination of ISDN call charges, need not be used in special rooms, better scalability and the potential for services of a better quality.

This dissertation attempts to understand the current JANET videoconferencing booking service and the facilities it provides, with a view to making recommendations about what requirements would be needed if it were deployed in an IP environment. The booking service will look easy to users, yet underneath, handle bookings of this complex web of resources.

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Chapter One

Introduction to JANET and Videoconferencing

1.0 Introduction

Videoconferencing is becoming an essential tool for collaboration [1] in the JANET community with a number of wide area video networks already established or being established. Videoconferencing over the wide area can be carried over a variety of network technologies e.g., ISDN, ATM (MAN), ATM (WAN), Leased Lines, IP Networks with the heart of many networks being video switches or Multipoint Control Units which enable multipoint videoconferencing.

1.1 JANET

JANET (the Joint Academic NETwork - www.ja.net) is the wide-area network created in 1984 to serve the needs of the higher education and research sector in the United Kingdom. The network and services are managed by UKERNA on behalf of the Joint Information Systems Committee (JISC). JANET is linked to other academic and commercial networks in the UK and abroad, forming part of the global Internet.

UKERNA offers the full range of user services over JANET including e-mail and newsfeeds, with supporting network services such as: Network Time Service, the Secondary Nameserver Service and Terminal Access Conversion Service.

Customer support staff can benefit from UKERNA's range of support services:

- JANET-CERT - The JANET Computer Emergency Response Team advises sites on network security matters and follows up security incidents;
- the JANET Operations Desk (JOD), based at University of London Computer Centre (ULCC), provides a single point of contact for customer support staff to report faults on any of the JANET services that are provided under the Service Level Agreements (SLAs)

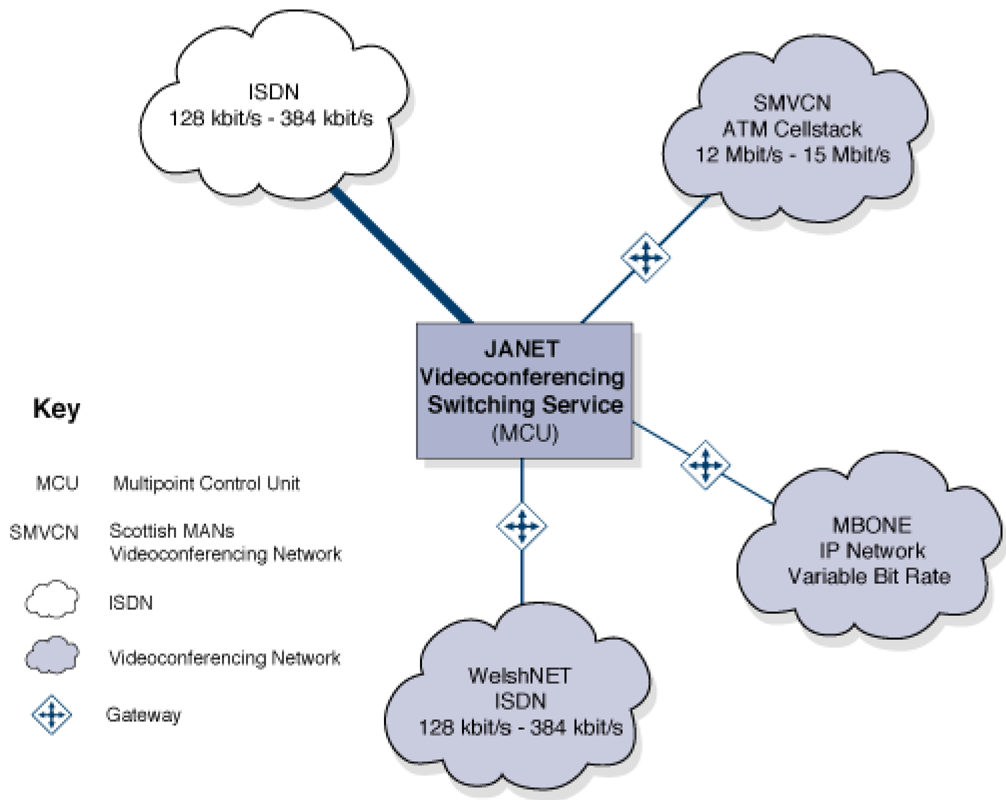
with the Joint Information Systems Committee (JISC); to minimise disruption regular maintenance is carried out during the JANET "At Risk" periods.

- JANET supports multipoint videoconferencing and UKERNA offers expert technical advice and training on equipment and techniques;
- traffic accounting services provide statistics on traffic over JANET and its external links
- an easy to understand view of the status and performance of the JANET network provided by netsight, a networking monitoring service

1.1.1 JANET Videoconference Service

The JANET Videoconferencing Services (JVCS) [2] allows JANET institutions to take part in multipoint videoconferences. The service is funded by the JISC, and managed and developed by UKERNA through the JANET Videoconferencing Management Centre based at the University of Edinburgh.

The JVCS consists of a national switching service for ISDN (Integrated Services Digital Network) based videoconferences, called the JANET Videoconferencing Switching Service (JVCSS); a videoconferencing service for Higher Educational Institution’s (HEIs) connected to the four Scottish MANs (Metropolitan Area Networks) that operate over the MAN ATM infrastructure in Scotland, called the Scottish MANs Videoconferencing Network (SMVCN); and IP videoconferencing capability operating over the JANET IP multicast infrastructure. To allow interworking between these networks, and the University of Wales videoconferencing network, WelshNET, gateways have been provided as part of the service. The topology diagram below illustrates the key components of the JVCS.



1.1.2 JANET Videoconferencing Switching Service

The JVCSS provides multipoint videoconferencing at speeds of 128 kbit/s (ISDN 2) and 384 kbit/s (ISDN 6) over ISDN. The service allows:

- Access to multipoint videoconferencing at ISDN 2 or ISDN 6.
- Access to multipoint videoconferencing with venues on the following videoconferencing networks:
    - Scottish MANs
    - WelshNET
    - MBONE

- All members of the UK Higher Education and Research community access to a central booking facility.
- Access to low cost multipoint videoconferencing. The core service is free of charge to institutions in the Higher Education and Research community. However, ISDN call charges are payable in connecting to the service.
- UK Higher Education Institutions and Research Council Establishments to utilise this service to connect with organisations outside the academic community to collaborate on joint projects.

To gain access to this service via ISDN, the following equipment is needed:

- Connection to the public ISDN network at either ISDN 2 or ISDN 6.
- A compatible COder DECoder (CODEC) with ISDN connectivity at ISDN 2 and/or ISDN 6. A CODEC is a piece of equipment which converts the audio and video signal, from the microphones and camera, into a signal which can be transmitted over the ISDN transmission lines. The CODEC also converts the received H.320 ISDN signal into a recognised signal for distribution to the loudspeakers and monitor.
- Suitable video cameras to provide the video feed to the CODEC.
- Suitable microphones to provide the audio feed to the CODEC.
- An echo canceller to stop distracting echo being heard during a videoconference. In the latest generation of CODECs, these are in-built.

Any JANET institution with suitable equipment can participate in a videoconference with up to 11 other institutions at 384 kbit/s (ISDN 6) or 35 other institutions running at 128 kbit/s (ISDN 2). To be able to exploit and connect to the services offered by the JVCSS, a venue must register with the Management Centre, through an online registration form. Each venue’s entry gives details of information such as names, telephone numbers and e-mail addresses of venue contacts as well as details of facilities available at the venue.

There are gateways to make communication possible between venues with different videoconferencing technologies. There is a gateway to each of the following networks:

Scottish MANs Videoconferencing Network

The Scottish Higher Education Institutions (HEIs) have an ATM based videoconferencing network running over their interlinked MANs. Most HEIs have a videoconferencing facility, either ISDN or ATM CODEC based.

C5C WelshNET Videoconferencing Network

The WelshNET Videoconferencing Network links the sites of Aberystwyth, Bangor, Cardiff, Lampeter and Swansea within the University of Wales. The network is mainly used for inter-collegial teaching sessions held across the University and is a *star network* where each site in the network has its own CODEC. All CODECs are connected to a central MCU located in the University of Cardiff via leased communication lines. All of the communication lines operate at 384 kbit/s. All WelshNET sites have a videoconferencing studio which is fully equipped with the necessary audio visual equipment (cameras, microphones, etc.). The WelshNET network is able to connect to the JVCSS via a direct ISDN connection between MCUs.

The JANET MBONE Service

The MBONE is currently implemented as a tunnelled network over the existing JANET IP service. By using IP packet multicasting the JANET MBONE provides a low cost transmission infrastructure for network multimedia applications.

1.1.3 The JANET Videoconferencing Booking System

With the exception of the multi vendor ISDN network, bandwidth is a limited resource, indeed for many institutions there is limited access to ISDN. Additionally, central video switching facilities, as well as many room based videoconferencing facilities at institutions are both shared resources. Given these resource restrictions, the need for scheduling and booking of the different components of a video network are essential.

Thus, the JANET Videoconferencing Service (JVCS) Booking System provides a web interface for registered users to request a JANET videoconferencing booking. It also provides details of videoconferencing facilities at each registered institution and a summary of existing bookings.

To request a booking, one must have a user ID and password. Existing bookings may, however, be browsed without the need to have a user ID and password.

The Booking System only contains details of conferences that are managed by the JANET Videoconferencing Services Management Centre. These include:

- Multipoint conferences
- Conferences gatewayed to other videoconferencing technologies.
- All conferences which include a SMVCN venue.

The booking system does not cover point to point ISDN based conferences. At present most JVCSS venues can only hold one conference at any one time. To run multiple, simultaneous, videoconferences at one venue, the venue will need to have multiple studios and multiple CODECs.

At present, each venue implements its own local booking system since it is impractical to suggest a standard booking system, with available facilities varying from venue to venue.

1.2 Objectives of the thesis

This study will set out to achieve the following objectives;

a) understand the current JANET VC Booking service and the facilities it provides

b) with reference to an example of a booking service (e.g. Welsh Video Network) that aims to use IP in the near future, extract requirements implied and the network arrangement, for IP-based videoconferencing

c) from a) and b) above, identify gaps in the current booking system, and propose some potential approaches

d) define a test strategy that others should adopt if they were to test this system

e) make recommendations for a new booking system

1.3 Methodology

In achieving the above set of objectives, the approach outlined in the following paragraphs will be adopted.

In order to make any meaningful recommendations regarding the present booking system, it's workings and aims will have to be understood clearly. Coupled with this, it is absolutely necessary to understand the whole phenomenon of videoconferencing, in general terms, from a non-layman point of view. This knowledge will be acquired from various sources including; the JANET and other related web sites, reviewing textbooks and works by other individuals relating to this area and through having discussions with my supervisor and other academic staff (of the Department of Computer of Science, UWA).

Understand the nature and technology of Internet based H.323 videoconferencing. With improvement in Information Technology and the growth of the Internet in global size and bandwidth, H.323 videoconferencing is fast becoming feasible. Resources needed for IP-based videoconferencing need to be understood, as well as the standards developed for this area.

With an understanding of H.323 videoconferencing and what resources need to be booked, desirable characteristics of the new booking system will be outlined and quantified. When these characteristics are matched against the present booking system, deficiencies in the system can then be identified.

Make a survey of the different components (identified in the previous paragraph as desirable for a booking system that will adequately support the booking of resources for H.323) on offer by different vendors dealing in the area of videoconferencing. The performance of each of these and how well they inter-work will be evaluated, so that components that best combine performance and inter-working will constitute the new booking system.

The selected mix of components will then be tested rigorously to assess their performance under different circumstances, say at different bandwidths, different levels of traffic (congestion) on the web. This may also include observing how the system recovers from, say a crush. The end result of this will be a system that meets the characteristics outlined above.

1.4 Structure of the thesis

From consultations carried out by the Quality of Service Think Tank [3], a strong demand from users (of the JANET) and their representatives for a well supported, Internet-based videoconferencing service was confirmed.

The main obstacles to videoconferencing over the Internet have been transmission bandwidth, packet loss, delay (latency), delay variation (jitter) and machine performance. The Internet is notoriously unpredictable when it comes to transmission performance. Heavy traffic load and internal transmission problems can cause delays that are beyond anyone's control.

As the Internet has continued to grow in global size and bandwidth, and as computer technology increases in speed and drops in price, coupled with better audio and video compression schemes, videoconferencing over the Internet is fast becoming more feasible.

This thesis explores the requirements and consequently implementation of a booking system, entailed by videoconferencing over the Internet. Of interest are what would be involved in such a booking, what resources would users need to book and the ability for the system (booking) to look easy to users, yet underneath handle the bookings of this complex web of resources.

The thesis is divided into six chapters, the first of which serves to introduce the JANET network and services, with emphasis on the videoconferencing services in general and the booking service in particular.

Chapter Two explores what Internet-based videoconferencing is all about, what standards exist in this area, what resources are required for such a videoconference and how do the components operate.

Chapter Three takes a look at the current JANET Videoconference Booking System (JVBS) and it's workings. This will serve as a first step at understanding and identifying gaps that exist in the JVBS.

Chapter Four covers a very important aspect that must be considered when data generated by applications that are in the real-time domain, like videoconferencing. The aspect of guaranteeing a Quality of Service (QoS) is key to a successful videoconferencing session. This chapter investigates issues involved in providing QoS and their possible shortcomings.

Given all aspects investigated in the preceding chapters, **Chapter Five** sets the way forward for requirements and implementation of a booking system in an IP environment.

While **Chapter Six** defines a test strategy that others should adopt if they were to test that the new booking system actually meets the stated requirements, **Chapter Seven** finally provides the conclusions and recommendations.

1.5 References

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Chapter Two

Internet-based Videoconferencing

2.0 Introduction

Internet videoconferencing is technology that allows the transmission of digital conversations and video pictures between computers connected on a network. This can be a Local Area Network (LAN), a Wide Area Network (WAN) or the Internet.

With the growth of the Internet in global size and bandwidth, and as computer technology increases in speed and drops in price, Internet videoconferencing is increasingly becoming more feasible. IP-based Videoconferencing frees implementation from the drawbacks of traditional transmission over dedicated rate networks (e.g. ISDN phone lines). These include the media being expensive, in the form of equipment and call charges, and being tied to a limited number of locations with special equipment and lines. IP-based transmission also opens up to more ubiquitous access, though network bandwidth plays a critical role.

2.1 Standards for IP Videoconferencing

In today's world of networked computers and the Internet, there is a critical need to maintain open architecture systems and standards. With video, voice and data all being sent down a common cable, each going to different (but possibly the same) destination, standards have been developed that allow the compression and transmission of this information.

Some of those standards are discussed below.

2.1.1 MBONE

The Multicast BackbONE (MBONE) is a global interconnected network of IP Multicast capable networks [1]. IP Multicasting is a technology that is better suited for broad-based video communications and has the ability to stream audio and video to hundreds or thousands of participants simultaneously, without complex configuration management requirements. In IP

Multicast, datagrams (UDP packets) are transmitted to a set of hosts through a single IP destination address. Each member of the set, known as a host group, then picks up the datagrams from the host-group address. Because IP Multicasting uses a virtual IP destination address to reach the host group, the sender does not need to know the connection details of, or have a direct connection to, all the members. Unicast, on the other hand, requires that the identification of each connection be known.

While the Unicast method would need to send multiple copies of data [2], one copy for each receiver, IP Multicast sends only one copy for all of the receivers at once. Thus multicast transmission reduces traffic load significantly, allowing multipoint conferencing to occur over any existing network infrastructure, typically with little extra cost or replacement of equipment. Bandwidth is more efficiently utilised, resources are saved, and datagrams are transmitted with less processing.

The MBONE allows multicast packets [3] to travel through routers that are set up to handle only unicast traffic. Software that utilises the MBONE hides the multicast packets in traditional unicast packets, through a scheme known as **tunnelling**, so that unicast routers can handle the information.

The MBONE, a loose confederation of sites that currently implement IP Multicasting, is thus a virtual network layered on top to the global Internet.

2.1.2 VRVS

The ``Virtual Room Video System'' (VRVS) has been developed since 1995 in order to provide a low cost, bandwidth- efficient, extensible means for videoconferencing and remote collaboration over networks within the High Energy and Nuclear Physics (HENP) communities [4].

In comparison to low-bandwidth commercial systems, videoconferencing for HENP (and some other scientific fields) requires higher resolution for several purposes, including shared documents, viewing remote presentations, and collaborative work. Hence LBNL (Lawrence Berkeley National Laboratory, Washington DC), Caltech, HEPNRC, and other members of the

Internet Community in the U.S. and Europe, have developed their own UNIX-based (and more recently PC-based) systems to meet the ongoing needs. In the longer term, these systems will adapt to, and/or supplement the new international standards for videoconferencing over wide area and local area networks as they emerge on the open market.

The Caltech/CERN-developed system, based on the LBNL protocol suite, provides low bandwidth videoconferencing with a high degree of interactivity (10-15 frames per sec in a bandwidth of 100-200 kbps and relatively high resolution), and gateways between IP-based and ISDN-based conferencing.

2.1.3 Access Grid

The Access Grid (AG) consists of large-format multimedia displays [5], presentation, and interaction software environment; interfaces to grid middleware; and interfaces to remote visualisation environment. With these resources, the AG supports large-scale distributed meetings, collaborative teamwork sessions, seminars, lectures, tutorials, and training. Informal (as well as formal) discussion for project collaboration, teaching, or other similar purpose can be supported by linked "spaces" – coffee lounge to lecture theatre – which are available at all times (though some sessions may be booked for specific purposes). In general, video streams from all participating sites are projected on a wall at each site, as are screen projections of shared applications.

The AG design point is group to group communication as opposed to desktop to desktop based tools that focus on individual communication. The AG environment enables both formal and informal group interactions. Large-format displays integrated with intelligent or active meeting rooms are a central feature of the AG nodes.

An AG node involves 3 – 20 people per site. AG nodes are "designed spaces" that support the high-end audio/video technology needed to provide a compelling and productive user experience.

2.1.4 The H.323 standard

The H.323 standard defines multimedia conferencing over packet-switched, generally IP-based, networks such as LANs/WANs and the Internet [6]. H.323 is intended for packet-switched networks with non-guaranteed bandwidth such as ethernet, token ring, FDDI, TCP/IP and IPX/SPX, with channel capacities up to 6/16 Mbps [7]. It covers both point-to-point and multipoint conferencing. The H.323 suite of communication standards was defined by the International Telecommunication Union (ITU) to provide a framework for interoperability between videoconferencing clients from a variety of different vendors. Thus, communications products offered by different vendors can now work together.

Under specification H.323, several components for real-time multimedia communications are defined. These include:

- **Terminals**: H.323-protocol terminals [8] are client endpoints that provide real-time, two-way communications. In order for a terminal to be compliant with the H.323 standard, it must support voice communications and can optionally support video and/or data.
- **Gatekeepers**: These are network-level administration servers that provide call control services to H.323-protocol endpoints, such as address translation, admissions control, bandwidth control, and zone management. An endpoint can be a terminal, gateway, or multipoint control unit.
- **Gateways**: Gateways allow interoperation of H.323 systems with other audio/video conferencing systems on integrated services digital networks (ISDN), plain old telephone systems (POTS), asynchronous transfer mode (ATM), and other transports.
- **Multipoint Control Units (MCUs)**: MCUs provide support for multipoint conferences between three or more endpoints.

Discussions in this thesis will focus on the H.323 standard, being the basis for interoperability for voice, video, and data conferencing implementations over IP.

Real-time H.323-based conferencing over IP networks provides the following key benefits:

- **Interoperability**

  The H.323 standard provides essential interoperability between communication end points (H.323 terminals, gateways, and MCU), thus making it possible for them to communicate with each other by means of H.225.0 signalling and H.245 control protocols.

- **The ability to integrate multimedia communications directly with other IP applications**

  The H.323 standard provides clear and simple architecture for optional data conferencing support through the T.120 series of standards that address point-to-point and multimedia data conferences. Thus making it possible for users to share applications, mark up shared whiteboard documents, and perform file transfers during the course of H.323-based conference.

- **Bandwidth Management**

  H.323 provides bandwidth management functions in the Gatekeeper to address congestion that would result in a network from the bandwidth-intensive nature of voice and video traffic. The gatekeeper achieves this function by approving or rejecting requests for call setups, and thus ensures that critical traffic is not disrupted.

- **Multicast support**

  An IP Multicast enabled H.323 terminal sends a single media stream to the IP network, and the networking equipment, i.e multicast enabled switches and routers, duplicate and distribute copies to every other H.323 terminal that is registered for the particular conference.

- **Scalability**

  The H.323 standard takes advantage of increased hardware performance and faster connectivity to improve software performance. A user can upgrade to a faster network connection or faster CPU and get a better videoconferencing experience from existing H.323-protocol software.

- **Long distance savings**

  Using the H.323 standard, long distance videoconferencing calls are placed over fixed-cost IP networks, as opposed to dial-up ISDN services with incremental usage charges.

- **Protection of investment in legacy conference systems**

  Conference systems that reside on H.320-based networks can now conference with H.323-based systems. Gateways that go from H.320-based environments to H.323-based environments bridge ISDN (circuit-switched) and IP (packet-based) networks.

- **Industry support**

  H.323 enjoys the support of many data networking and telecommunications companies.

- **Leveraged investment in existing corporate network infrastructure**

  Traditional H.320-based conferencing systems require dedicated ISDN lines for each conference room or office. The H.323 standard can be deployed wherever Internet Protocol (IP) is supported.

- **Convenience to the user**

  The H.323 standard can be integrated cost-effectively into users' existing desktop computers, as compared to the H.320 standard which traditionally meant expensive room-based systems, requiring the user to go to the technology.

- **Increased productivity**

  Since H.323 can be deployed on users' existing desktop computers, these desktop systems can be used throughout the day for general business purposes when users are not participating in videoconferencing calls. Dedicated H.320-based conferencing systems sit idle when there is no videoconference session.

2.2 Resources needed for an IP-based Videoconference

2.2.1 People who want to meet

A set of participants wishing to videoconference discuss by traditional methods of telephone, email or face to face, when they wish to meet and at which venues. Among other issues, they also decide upon who amongst themselves should be the conference organiser.

2.2.2 Videoconferencing terminal stations (VC clients)

Most, if not all of the H.323 systems, can operate on a general-use platform such as a personal computer. This ability allows the H.323 system to scale with underlying processing power. Clients should also scale performance based on the available bandwidth and should be set with proper audio and video CODECs.

Clients can be of same or different vendors as long as H.323 compliance is verified. Mechanisms should be provided for, at the client platform for fault tolerance and error resilience. This is aimed at addressing the inconsistencies of networks with best effort traffic such as the Internet (i.e. no guaranteed Quality of Service).

2.2.3 Peripherals, including cameras, microphones and speakers

It is important to consider equipment that will upgrade and/or scale to the near future without the requirement of replacing it at each step.

Audio requirements [9]

- Hands-free or via telephone handset
- Audio input
  - Microphone or camera microphone
  - Telephone handset or headset
- Audio output
  - Loud speakers or external output
  - Telephone handset or headset

- Software selection and control of audio inputs and outputs
- Sound coding in G.711 (56 or 64 Kbps)
- Sound coding in G.723.1 (5.3 or 6.4 Kbps)
- Lip synchronization with sound
- Echo cancellation and noise suppression

Video requirements

- Camera

  The camera should be colour balanced effectively and checked for efficient operation of lens focus, zoom, iris, pan, and tilt where appropriate
- Video input for PAL or, NTSC
- Video display
- Full-motion video compression

  H.261 CIF (352 x 288) and QCIF (176 x 144)

2.2.4 Workstations with IP addresses

2.2.5 High speed network connection (at least 128Kbps.)

2.2.6 Gatekeepers - Optional

A gatekeeper provides call control services to the H.323 endpoints [10] such as address translation, admission control, bandwidth control, and zone management. It may also provide other optional functions such as call authorisation and call accounting.

Through allocation by administrators of the desired amount of bandwidth for conferencing traffic to different gatekeeper zones, they are able to make real-time multimedia conferencing available to many users without fear of overloading the network. When an H.323-protocol call is made, the gatekeeper accepts or rejects it based on current network load at the time.

2.2.7 Multipoint Conferencing Unit (MCU) – Optional

Multipoint conferences involve calls between three or more parties. During these conferences, call control and media operations are considerably more complex than during a simple point-to-point conference. Thus the coordination and notification of participants entering and leaving a conference along with the marshalling of the media streams requires the presence of a Multipoint Control Unit (MCU) or using multicast. Both approaches lead to dramatic savings in bandwidth over a wide area network or the Internet.

MCUs can be implemented in either software or hardware. Hardware implementations tend to be more expensive and are likely to contain a variety of proprietary components but are likely to be faster and more reliable. Software implementations are more portable, more flexible, and less expensive but may suffer performance issues due to their reliance on the operating system and resources of the computer they are running on. Software-based MCUs rely on host computers that must be fast enough to keep up with all of the video streams in the conference and that the load on the computer increases with the number of people in a conference.

However, both implementations can be connected together to allow larger numbers of sites to be conferenced together simultaneously. A careful matching of performance requirements to cost variables should be combined with a broad comparison of available products within each implementation type before a final buying decision is made.

2.2.8 Security of media streams in an unsecured environment such as the Internet

The H.323 system can utilize underlying security protocols such as Internet Protocol Security (IPSEC) or Transport Layer Security (TLS) as established in the IETF, to provide services of authentication (which can be used for authorisation), privacy, and integrity.

2.2.9 Gateways

A gateway operates as an endpoint on the network that provides real-time, two-way communication between H.323 terminals on the packet-based network and other ITU terminals on a switched-circuit network, or to another H.323 gateway. It also performs call setup and clearing on both the network side and the Switched-Circuit Network (SCN) side, translation between video, audio and data formats.

2.2.10 Firewall design that supports H.323-protocol traffic

Firewalls are designed to protect an organisation's network from unwanted computer access. If a firewall is setup, then it is desirable that it supports H.323-protocol traffic. The firewall can be programmed to pass H.323-protocol multimedia traffic and configured to have enough bandwidth to carry the same.

2.2.11 Quality of Service (QoS) measures for LANs

- For Shared Ethernet, LAN segments should not be loaded to more than 35 – 40% of capacity in order to accommodate spikes in usage
- Upgrading from a Shared Ethernet LAN to a Switched Ethernet LAN would be recommended. An Ethernet switch isolates traffic between different users so that heavy demand from one user doesn't slow down performance for others. A more cost-effective way to add bandwidth is to spread LAN users across multiple Ethernet switches.
- Use should be made of routers that give audio and video priority over non-real-time data such as e-mail and file transfer.
- Particular attention should be paid to the amount of latency introduced by each component on the network. In order to maintain quality between sender and receiver, and to ensure that audio and video data do not lose relevance to the receiver due to untimely delivery, low latency routers, switches and other network devices should be used.

2.2.12 Bandwidth Broker

A Bandwidth Broker (BB) is a software entity that manages resources for IP QoS Services [11] supported in a network and used by customers of the network services. It is responsible for internal and external admission control decisions according to a policy database. Based on such decisions, it configures any routers within the domain and is also responsible for negotiating with BB's from neighbouring domains.

Since a BB touches on a number of functions in the network, including network, policy control and configuration management, these functions may in fact be obtained as services from other nodes implementing them (e.g. a Gatekeeper), rather than these functions being implemented in the BB itself. It may have interfaces to other functional entities in the network. Alternately, these functions may be implemented or packaged with the BB.

2.3 How a typical H.323 videoconference would operate

The steps below provide an overview of the basic operation of an H.323 endpoint in a point-to-point conference, with a gatekeeper.

- **Discovery and Registration**

  Before a conference starts, endpoints (also called H.323 terminals or clients) look for one of the gatekeepers by multicasting a Discovery Request [12], and if present, register their alias names with it using the Registration, Admission, and Status (RAS) protocol. Registering alias names with the Gatekeeper allows endpoints to call each other using user-friendly addresses such as e-mail, etc., rather than the transport addresses.

The Discovery and Registration procedure is valid until the gatekeeper indicates otherwise.

- **Address Resolution**

  Communication starts when user initiates a call e.g. by dialling on an IP telephone or on a H.323 compliant software client. An endpoint generates a message to notify the gatekeeper that a call must be placed to a particular destination. Since calls can be placed with user name, extension, or any other alias, the gatekeeper uses its knowledge about the configuration to return the IP address of the destination. In addition to the IP address returned, this admission message carries the initial bandwidth the endpoint requires for the duration of the conference. If during H.245 logical channel negotiation, an endpoint requires more bandwidth, it issues a bandwidth request to the gatekeeper. A request which the gatekeeper may accept or reject. Other H.245 control messages may be exchanged between the endpoints to change media formats, request video key frames, change the bit rate, etc.

After receiving the necessary information, the initiating endpoint places a call to the destination endpoint that can also be a gateway to the switched telephone network. For the H.323 terminal to perform these functions, it must support H.225.0 signalling, H.245 call control, RAS, and voice and/or video compression.

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Chapter Three

The existing JANET Videoconference Booking System

3.0 Introduction

With the exception of the multi vendor ISDN network, bandwidth is a limited resource, indeed for many institutions there is limited access to ISDN. Additionally, central video switching facilities, as well as many room based videoconferencing facilities at institutions are both shared resources. Given these resource restrictions, the need for scheduling and booking of the different components of a video network are essential.

The current booking system [1] supports the JVCSS and SMVCN services. The online booking system is managed by the Videoconferencing Management Centre with videoconferencing resources being booked on a first come first served basis. Registered institutions are required to have a booking contact at their site who will request bookings and it is only these contacts that can request bookings. A booking specifies a set of resources to be allocated at a certain time for a certain period.

The booking system contains information on the conferences that are managed by the JANET Videoconferencing Management Centre. These include: multipoint conferences, conferences gatewayed to other videoconferencing networks, all conferences (multipoint and point to point) on the SMVCN. If a venue's facilities (studio and/or equipment) are unavailable due to maintenance or the facilities are being used for other purposes, then it is the responsibility of the booking contact concerned to book out the facilities that they are responsible for within the booking system. Only registered users of the booking system may request a booking to be made. The period of time, and the resources required, are presented to an on-line booking system via a WWW based form.

3.1 Current hardware and software

The current booking system runs under Red Hat Linux and uses a SQL database (postgres) to maintain all the system data (booking parameters, user information, venue resources etc). The HTML pages call CGI functions written in C to interact with the SQL database to populate the HTML forms and to process and verify the data submitted back by the form. The system also performs network resource and bandwidth validation when a request for a videoconference is made.

3.2 Booking Procedure

The booking system only covers the central JVCSS facilities and venue CODEC [2], not the rooms, etc at each venue. It has been more difficult to ensure that the local bookings are satisfactory because each centre has its own booking procedures for local resources, and it is extremely difficult to establish a global system.

Gateways are automatically booked when venues on different networks are booked in the same videoconference. Thus, no additional booking is required.

3.2.1 Step-by-step procedure to booking a videoconference

The steps outlined below are to be taken by both the user and the local booking contact:

- User agrees with the others to take part in a meeting on the dates and times of the meeting, the venues to be involved and on the conference organiser
- Check if the dates and times agreed upon are free by consulting "Summary of Current Bookings" at http://www.jvcs.video.ja.net/videoconf/list_bookings.shtml
- Consult http://www.jvcs.video.ja.net/cgi-bin/venue.cgi?mode=list for facilities in the list of registered venues and decide on what will be needed at each venue
- With the information below, the local booking contact (http://www.jvcs.video.ja.net/cgi-bin/site.cgi?mode=list) is approached to request a booking for a conference

- the date(s) of the meeting(s);
- the start and end times;
- the venues, rooms and equipment required;
- the name of the meeting (a series can also be named);
- the name and contact details of the organiser;
- your name and contact details and the name and contact details of one person at each of the other venues involved;
- whether you do not wish "Audio Switched"
- whether you do not wish "Continuous Presence"

When three or more venues are involved in a videoconference there is a choice as to the type of conference set up available, the choice being between "Audio Switched" and "Chairman Controlled".

When availed with the above information, the local booking contact goes ahead to:

- arrange for the reservation to be made as a booking request on the booking system using http://www.jvcs.video.ja.net/videoconf/index.shtml
- contact the local booking contact at each site involved to ensure that they make the necessary local bookings at their site
- inform the conference organiser (user) that all is ready, when all sites confirm that local facilities have been booked.

The local booking contact could also make a booking by sending a booking request by e-mail. A standard form, shown below, is used in this case. To get a copy of the email form, an e-mail message without any text is sent to JVCSform@ed.ac.uk.

3.2.2 An example of a typical E-mail Booking Form

N.B. FOR CONFERENCES INCLUDING M-BONE PARTICIPANTS PLEASE USE THE WWW BOOKING FORM AT http://www.jvcs.video.ja.net/videoconf/index.shtml

---

Conference title :
Start date : (dd/mm/yy)
Start time : (00:00)
End date : (dd/mm/yy)
End time : (00:00)

Contact Information: (Information on Contact person(s))
Name:
Telephone Number :
E-mail address :

WelshNet sites : (Site(s) name(s))

Scottish MAN sites : (Site(s) name(s))

ISDN sites : (Site(s) name(s))

ISDN channel type request: [2B] [6B] [6Bbonding] [Other]

Continuous presence: [YES] [NO]

User Supplied Notes:

3.2.3 An example of a typical confirmation received, of a booking made

+++

From: JANET Videoconference Booking System

[mailto:vidconf@jvcs.video.ja.net]

Sent: 08 August 2001 09:17

To: JVCS@aber.ac.uk

Subject: Videoconf AU411137 - 'SuperJANET Multicast project meeting'

The following booking has now been confirmed -

Conference Title: SuperJANET Multicast project meeting

Booking Ref: AU411137

Purpose: Administration/Management Meeting

Starting at: Wed 12 Sep 10:00:00 2001 BST

Ending at : Wed 12 Sep 12:00:00 2001 BST

Booked by: ukerna-jo (Jo McInnerny, Primary Booking Contact, UKERNA)

<jvcs@ukerna.ac.uk>

Conference Organiser: Jeremy Sharp

Email: j.sharp@ukerna.ac.uk

Telephone: 01235 822 259

Conference venues:

ABER-DILS_125 (DILS 125) [6B_BONDING]

UKERNA-ATLAS-G56.57 (UK Education and Research Networking Association)

[6B_BONDING]

ULCC (University of London Computer Centre) [6B_BONDING]

Principal Attendees at the above venues:

UKERNA (Jeremy Sharp), ULCC (Rob Evans), Aberystwyth University (Dave Price)

==

ISDN Numbers for venues:

ABER-DILS_125: 087 0085 4414

+++

After such a confirmation, the room under consideration is then booked as "unavailable" for other C5C use and the system then sends out the following email

+++

Date: Wed, 8 Aug 2001 14:46:30 +0100 (BST)

From: Aberystwyth Booking User <c5cab@swansea.ac.uk>

To: dap@aber.ac.uk

Subject: C5C New Booking

C5C Video Conferencing Booking System

SITE UNAVAILABILITY

Made on......: 8/8/2001

Conf. date...: 12/9/2001

Start Time...: 10:00

End Time.....: 11:59

Description..: Tandberg Super Janet Meeting

Contact......: Dave Price

Tel. No......: 01970 622428

Email........: dap@aber.ac.uk

Sites involved in conference

AB - Aberystwyth

*** End of Message ***

==

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Chapter Four

Quality of Service and Bandwidth Broker

4.0 Introduction

The Internet is a network of networks, a mesh of various transmission media, with a wide range of bandwidth capacity and latency characteristics. Link status between two hosts across the Internet can vary widely from one millisecond to the next. Network application traffic is busty, and with many applications sharing the same network links at the same time, transient congestion is often the result.

The Internet offers no network prioritisation, with all data being treated equally. Data is transmitted on a best-effort basis and guarantees for any sort of network resources is not provided. The Internet [1] does not undertake to deliver (packets may be dropped to relieve congestion), to deliver in order (packets may not all follow the same route, and routes are not necessarily of the same length), to deliver undamaged (bit errors are neither detected nor corrected by the network), or to deliver to any timescale (either in terms of end-to-end delay or delay variation between one packet and the next).

The above behaviour is a limitation for applications, such as conferencing and telephony, that demand high data throughput capacity (bandwidth) and have low latency, jitter, and loss requirements when used in two-way communications.

4.1 Quality of Service

Quality of Service (QoS) means providing a consistent [2], predictable data delivery service. The focus being on providing predictable service during periods of congestion. QoS is qualified by a number of characteristics;

- Minimising delivery delay

- Minimising delay variation
- Providing consistent data throughput capacity

QoS does not create bandwidth (bandwidth is a finite resource) but instead manages bandwidth according to application demands and network management settings [3].

Essentially, two types of QoS are available:

4.1.1 Integrated Services (IntServ)

IntServ has three classes of services defined that, if supported by the routers traversed by a data flow, can provide the data flow with certain QoS commitments [4]. IntServ provides for *per-flow management*, in which each router perceives the individual traffic flow and allocates network resources to each flow. The level of QoS provided is programmable on a per-flow basis according to the requests from the end applications. These requests can be passed to the routers by network management procedures or, more commonly, using a reservation protocol such as RSVP. The requests dictate the level of resources (e.g., bandwidth, buffer space) that must be reserved along with the transmission scheduling behaviour that must be installed in the routers to provide the desired end-to-end QoS commitment for the data flow.

4.1.2 Differentiated Services (DiffServ)

By recognising that most of the data flows generated by different applications can be ultimately classified into a few general categories (i.e., traffic classes), the differentiated services (DiffServ) architecture aims at providing simple, scalable, service differentiation. DiffServ attempts to push the workload from the center of the network to the edge of the network. With DiffServ, hosts and routers at the edge of the network classify packets into a limited number of service (these services can be either end-to-end or intra domain) categories, then pass the packet into the core of the network. Core routers extract the packet label and put the packet into a pre-established priority level. This process requires substantially fewer resources per-flow than IntServ.

4.2 Bandwidth Broker

The Bandwidth Broker (BB) is a software entity [5] that manages resources for IP QoS services in a Differentiated Services network. It is responsible for allocating different services to users as per their requests and the existing service policy in the network, and for configuring the network routers with the correct forwarding behavior for the defined service. As one of the mechanisms to address network QoS, a BB keeps track of the current allocation of marked traffic and interpreting new requests in the light of the policies and current allocation.

The BB will be in charge of both the internal affairs and external relations regarding resource management and traffic control. Internally, a BB may keep track of QoS requests from individual users and applications, as necessary, and allocate internal resources according to the domain's specific resource usage policies. Externally, a BB will be responsible for setting up and maintaining bilateral service agreements with the BB’s of neighbour domains to assure QoS handling of its border-crossing data traffic.

A BB is associated with a particular trust region, one per domain. A BB has a policy database that keeps the information on who can do what, when and a method of using that database to authenticate requesters. Only a BB can configure the leaf routers to deliver a particular service to flows, crucial for deploying a secure system.

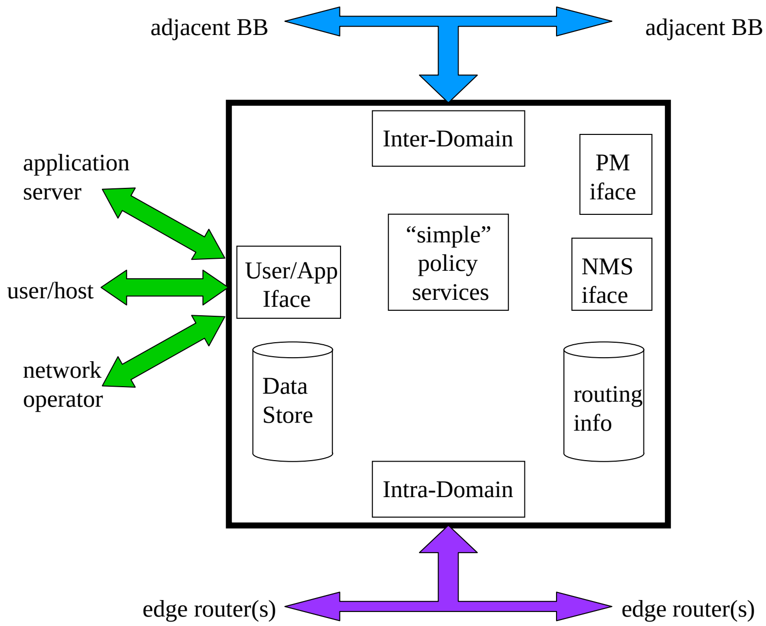
When an allocation is desired for a particular flow, a request is sent to the BB. Requests include a service type, a target rate, a maximum burst, and the time period when service is required. The request can be made by a user or it might come from another region's BB. A BB first authenticates the credentials of the requester, then verifies there exists unallocated bandwidth sufficient to meet the request. If a request passes these tests, the available bandwidth is reduced by the requested amount and the flow specification is recorded.

The BB configures the appropriate leaf router with the information about the packet flow to be given a service at the time that the service is to commence. This configuration is "soft state" that the BB will periodically refresh.

4.2.1 Bandwidth Broker Architecture

A BB can have the following general components, which are illustrated in the figure below;

- User/application interface
- Bandwidth Broker Database.
- Bandwidth Broker router configuration client.
- Intra-domain Communication Interface
- Inter-domain Communication Interface
- Policy Manager (PM) Interface
- Network Management (NM) Interface



4.2.1.1 User/Application Interface

This interface will provide for the following functionalities:

- Manual user/network manager GUI

  The GUI will be used by the network operator to perform various tasks such as populate the BB database with simple policy rules, network topology and configuration information; to manually input BB service requests to the system; query the BB for existing service commitments, resource provisioning, etc; and as a platform through which errors, etc will be communicated to the operator.

- Requests directly from user/application on end host

  Host/user applications can directly make BB service requests to the system; or users can query the BB for existing service commitments (granted to them), security required to prevent users from viewing commitments made to other users.

- Requests from application servers

  This provides a server/gateway interface for requests from application servers (e.g., H.323) on behalf of hosts/users.

4.2.1.2 Bandwidth Broker Database

The database provides an API that is used by the server to interact with it for customized queries. It includes a mechanism for storing all data related to the functioning of the Bandwidth Broker within its trust region. The following are among the entities that a BB stores, which may be shared by other network components such as policy control and network management;

- Service Level Agreements (SLA)
- Bandwidth Allocation Requests (BAR)
- Mapping of Bandwidth Allocation to Code Points.

Entries are made to the database whenever a request from the command line interface or from a client is accepted.

4.2.1.3 Bandwidth Broker Router Configuration

The BB configures the leaf and egress routers in its domain according to the service level agreements and incoming bandwidth allocation requests from clients. It may require access to inter-domain routing information in order to determine the egress router(s) and downstream domains whose resources must be committed before incoming resource allocation requests (RAR) may be accepted.

Additionally, a bandwidth broker may require access to intra-domain routing information in order to determine the paths and therefore resource allocation information within the domain.

4.2.1.4 Intra-domain communication interface

This is for communication of BB decisions to routers within the bandwidth broker's domain in the form of router configuration parameters for QoS operation and (possibly) communication with the policy enforcement agent within the router. The BB keeps track of the DiffServ traffic that enters and leaves the domain across its boundaries, making sure that the bilateral agreements with adjacent domains are adhered to.

4.2.1.5 Inter-domain Communication Interface

This provides for a mechanism for peering (adjacent) BB's to ask for and answer with admission control decisions for aggregates and exchange traffic. Through such a mechanism, resources are provisioned and allocated resources at network boundaries between two domains. A bilateral service-level agreement (SLA) specifying the amount and types of traffic each side agrees to send and/or receive must be established on the boundary between two domains.

Signaling messages are sent between BBs of adjacent domains to request from the adjacent BB the necessary resources in the adjacent network, and to communicate information about the resources required on the links connecting the domains.

4.3 Glossary

Domain

In general, a collection of nodes (hosts, routers, etc) and a set of links connecting them. A domain, in the context of this chapter, refers to a collection of nodes and links that are under the control of the Bandwidth Broker.

Peer Domains

Domains which are adjacently connected

Edge Router (or Edge Device)

A network device, usually a router, which directly connects two or more domains. Although it has interfaces in more than one domain, it is controlled by only a single BB.

Intra-domain communication

Intra-domain communication refers to the protocol messages and control data that gets exchanged between a BB and the nodes (usually edge devices) within that BB's domain.

Inter-domain communication

Inter-domain communication refers to the protocol messages and control data that gets exchanged between BB's in adjacent domains.

Service Level Agreement (SLA)

A business agreement established between a service provider and a customer. A customer may be a user organisation (source domain) or another domain. An SLA typically contains such things as: terms and agreement regarding the type of service(s) the customer will receive, possibly how the provider will demonstrate to the customer that the service was provided, how payment will be made to the provider, actions that can be taken by either party if the service/payment is not received, etc.

Service Level Specification (SPS)

An SLS refers to the particular information relative to the BB and the network devices in order to support an SLA in that network. Information in an SLS is generally on the level of network ports, IP addresses, (aggregate) data flows, resources/bandwidth, etc. An SLS is typically applied at the endpoints of a link connecting adjacent domains and reflects traffic that will be sent from the source the upstream domain to the downstream domain.

Bandwidth Allocation Request (BAR)

Bandwidth Allocation Requests are used by clients to request portions of a service allocated to them by an SLA, for individual flows. The Bandwidth Broker, before granting the request ensures that allocation will not violate the limits of the SLA and also will not cause the aggregate quantity of that service to be exceeded in the trust region.

Resource Allocation Request (RAR)

A request for network resources (or service) from an individual user to the BB of that user's domain. If the request includes network resources for traffic to a destination(s) outside of the user's local domain, the admission control may be performed based on the SLS(s) in place with adjacent domains.

DiffServ egress node

A DiffServ boundary node in its role of handling traffic as it leaves a DiffServ domain.

DiffServ ingress node

A DiffServ boundary node in its role of handling traffic as it enters a DiffServ domain.

Downstream DiffServ domain

The DiffServ domain downstream of traffic flow on a boundary link.

Upstream DiffServ domain

The DiffServ domain upstream of traffic flow on a boundary link.

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Chapter Five

Requirements for the proposed booking system

5.0 Requirements

The current JANET videoconferencing provision is based on ISDN technology and ATM technology, which is used within the Scottish Higher Education community.

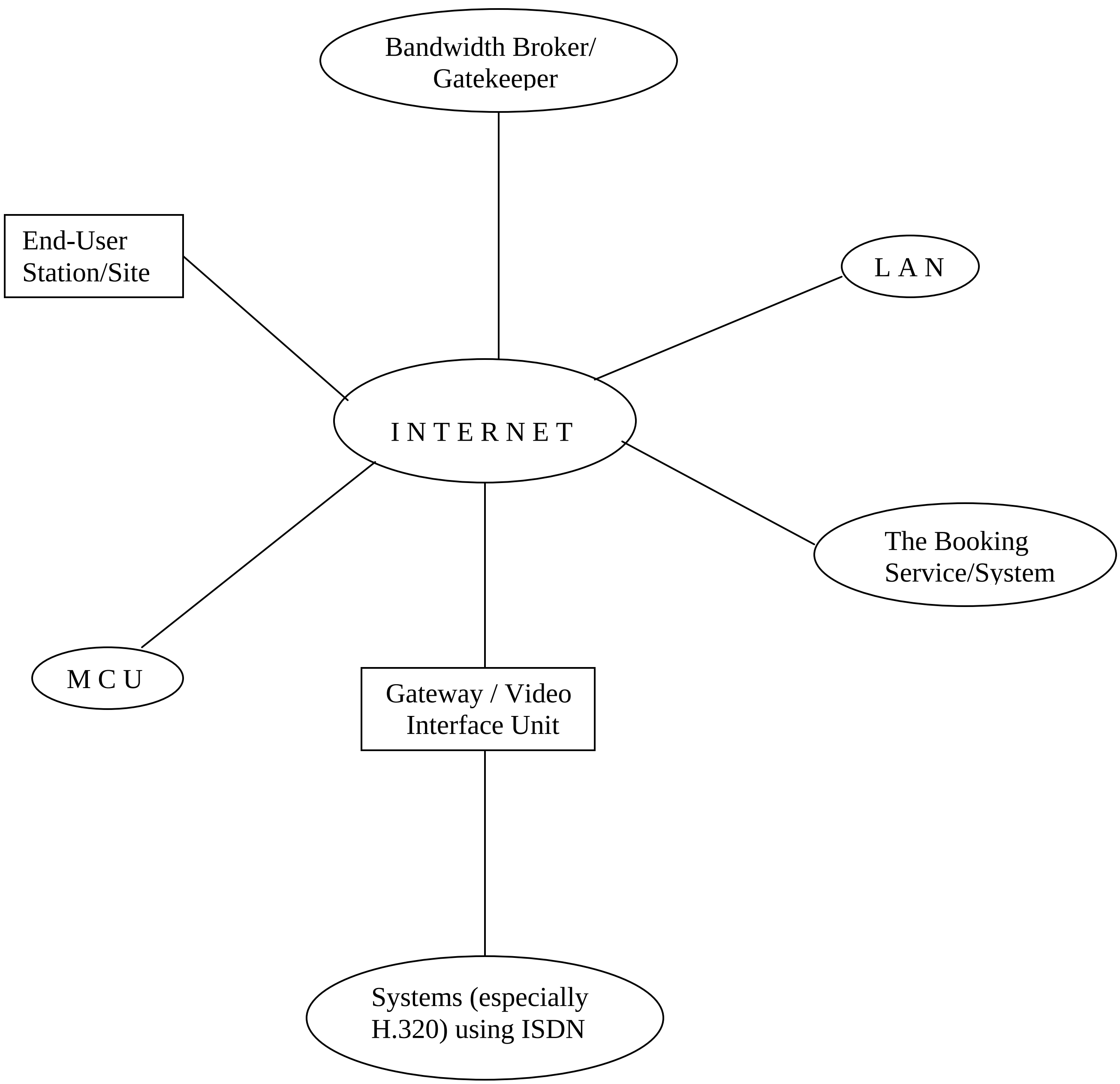
The proposed booking system will have the capability to recognize and handle booking requests initiated from the above technologies as well as those originating from users on IP-based networks i.e. H.323.

The booking system will be able to interact with among others, existing videoconferencing equipment and the bandwidth broker. Such interactions will take the form of scheduling and configuring MCU, gateways and gatekeepers; querying the bandwidth broker for availability of required bandwidth along the path of points that intend to videoconference.

The system will, in addition, be able to book any combination of conference resources such as all the hardware required to make a booking. Such hardware would include: Multi-point Control Units (MCU), video cameras, document readers, slide projectors, etc. This capability demands that the booking system maintains a repository of all hardware and attributes relating to them. When a new hardware becomes available, it's repository is updated. Similarly, the removal of any hardware for any reason, leading to that hardware's non-availability, should be reflected.

Once a venue or any videoconference resource is booked, the system books out this venue/resource so that subsequent bookings of the same are not possible. The system sends a log to the respective person(s) In-charge of these resources notifying them about the booking details. This can be achieved by way of email or fax. This, is particularly desirable for those venues which in addition to videoconferencing, host other activities such as lectures, meetings or television studios. Advance knowledge of such a booking would give ample time to those concerned to guarantee the availability of such a session.

5.1 Architectural view of the proposed solution



5.2 How the proposed solution would operate

In the likely event that a user accesses the booking service to make a reservation for a future conference, a series of queries is initiated and set into action. Firstly, on receiving this request, the service queries it's database to establish whether the requested time slot is available or not. In case this time slot is already booked, the user is accordingly informed about this collision and advised to try booking another time slot.

However, if the requested time slot is available i.e. not booked, the service queries the bandwidth broker (BB) for availability of the desired resources to meet the needs of the conference, along the path of the conference. The BB responds by querying routers along this path. A positive response is fed back to the booking service if resources, say bandwidth, are available. Its only then that the service confirms the booking to the user and then goes ahead to update it's database.

Similarly, the BB would notify the service if resources were absent or not sufficient, so that it would in turn notify the user (indicating the reason). In this latter case, no booking would be effected.

5.3 Components of the proposed solution

It should be noted that all of the equipment components are connected to the Internet [1], and not to each other. Equally important to note is the fact that none of the video or audio flows through the gatekeeper, even though it controls all the other equipment.

5.3.1 Bandwidth Broker/Gatekeeper

The BB is configured to receive and respond to requests from the booking service about resource availability between end-user points that wish to videoconference.

Since the BB touches on a number of functions in the network, including network, policy control and configuration management, these functions may in fact be obtained as services from other nodes implementing them (e.g. a Gatekeeper), rather than these functions being implemented in the BB itself. It may, however, have interfaces to other functional entities in the network. Alternately, these functions may be implemented or packaged with the BB.

The Cisco Multimedia Conference Manager (MCM) is one such product [2] that packages the functions of a BB and Gatekeeper under one "roof", so to speak, providing an essential element of management, security and QoS requirements for IP videoconferencing. It consists of an H.323 Gatekeeper and an H.323 Proxy.

The MCM makes it possible for H.323 traffic on the LAN and WAN to be limited as well as providing user accounting for records based on the service utilization. Additionally, it injects QoS for H.323 traffic generated by applications such as video conferencing, and provide a mechanism for implementing security for H.323 communications.

On the other hand, the Gatekeeper component of the MCM functions as a point of control for a variety of audio and video components that can be attached to an IP network; IP telephony devices, IP-PSTN Gateways, H.323 videoconferencing endpoints, H.323 Multipoint Control Units, etc.

A product similar to Cisco's MCM, PacketAssist™ [3], is offered by Visual Communications (VCON). The product has an in-built Adaptive Bandwidth Adjustment and QoS features which ensure that if there was a possibility of network congestion, the audio and video will be optimized so that the end user experiences the best quality possible.

5.3.2 Multipoint Control Unit (MCU)

An MCU is a server equipment that makes it possible for more than two (2) sites/endpoints to videoconference. All sites call into the MCU, which sends back to each site the video it has selected.

To ensure interoperation between IP and the current ISDN based JANET videoconferencing service, at minimum, two MCU's should be deployed, each supporting sites on one of the two technologies. The MCU's will be configured to support voice switched videoconferencing and be able to initiate calls to remote CODECs.

Polycom offers a Video MCU [4] with improved conference connection rates with optimal capabilities, T.120 data conferencing across mixed ISDN and IP networks. Features such as IP Continuous Presence and IP QoS are included.

Cisco offers the IP/VC 3510 MCU which combines video, audio, and data streams from multiple conference endpoints into one multi-location interactive session. When used in conjunction with the IP/VC 3520 and 3525 Videoconferencing Gateways, the IP/VC 3510 MCU can also bring one or more H.320 endpoints into the conference

5.3.3 The Booking Service/System

The booking system is essentially a database and as such, should conform to the acceptable principles of a good database management system. One such requirement is the ability to ensure system security and database integrity. Different users have different levels of access to the system and this should not be breached. Resources once booked, should be available for the duration of the conference, and nothing less or more.

Authorised users should be in position to make, modify or cancel bookings without any difficulty. However, amendments or cancellation of any booking is only applicable to those bookings whose date and time of maturity has not passed, and only by the authorised user who made the booking. In which case, any hardware which had been configured for the original date/time should be re-configured to reflect such changes.

The booking system will produce vital statistics for management and auditing purposes. These will include such information as the duration of a videoconference so that billing is made easier, a log of errors and how long they lasted, confirmation to the parties concerned about any bookings made, etc.

5.3.4 Gateway

A Gateway provides translation services between H.320 and H.323 networks to relay audio and video streams between circuit-switched ISDN and IP networks. In so doing, it enables users to videoconference with others via the LAN or the Public Switched Telephone Network (PSTN), regardless of location hence preserve legacy investment.

The Cisco IP/VC 3520 and 3525 Videoconferencing Gateways enable videoconferencing sessions of rates ranging from 64kbps to 768 kbps. These gateways also support a variety of call routing methods including interactive voice response (IVR), direct inward dialing (DID), multiple subscriber number (MSN), TCS4, or route to a default destination.

A similar product offered by RADVISION, vialIP videoconferencing gateway version 2.0 [5], provides complete IP and ISDN interworking functionality with features designed to provide high quality, high availability call management with enhanced packet handling and integrated network resource management.

5.4 References

1. Internet Videoconferencing: Coming to Your Campus Soon!
   Robert S. Dixon

   Educause Quarterly, Number 4 2000

2. Cisco Systems

   http://www.cisco.com

4. Visual Communications

   http://www.vcon.com/products/desktop/

5. Polycom Worldwide

   http://www.polycom.com

6. RADVISION

   http://www.radvision.com

Chapter Six

Testing and Evaluation

In order to establish that the proposed booking service conforms to its specification and will meet the needs and requirements of the JANET users who wish to videoconference, the following test strategy should be adopted.

6.1 Booking system

Through an on-line WWW-based form, the booking system database should initially be populated with information relating to (JANET) institutions that will have a need to videoconference. These registered institutions are required to have a booking contact at their site who will request bookings and it is only these contacts that can request bookings. A typical registration showing the required parameters is shown below.

Showing: institution:ABER	
Institution	ABER
Description	University of Wales, Aberystwyth
Primary Contact	aber-tom
Operations	is-vidconf@aber.ac.uk
Home Page	http://www.inf.aber.ac.uk/vidconf
In Service	True

At each of these registered institutions, details about sites at which videoconferencing facilities are located are then added to the booking system. An example follows.

Showing: institution:ABER , site:ABER-COMP-SCI	
Site Name	ABER-COMP-SCI
Description	University of Wales Aberystwyth, Computer Science Department
Primary Contact	aber-dap
Site Manager	aber-dap
Owning Institution	ABER
In Service	True

At each of the sites entered, say, the Department of Computer Science at the University of Wales, Aberystwyth, there may be one or more videoconferencing venues. These are added to the booking system, indicating what resources are available at each. The condition of each of these resources should also be specified i.e. whether in good working condition or not. It is desirable that these sites should support call speeds of at least 128 kbits/s. The table below shows a list of resources available at one of the venues at the University of Abertay, Dundee.

Resource	**Description**	**Venue**
CAMERA_1	Fixed position wide shot JVC TK1281	ABERTAY-BELLST-3014
CAMERA_2	Remote control camera with 6 preset positions - Sony EVI D31	ABERTAY-BELLST-3014
CAMERA_3	Manually operated camera - Sony DXC M7	ABERTAY-BELLST-3014
DATA_PROJECTOR	Epson Data Projector and pull down screen	ABERTAY-BELLST-3014
DOCUMENT_CAMERA	JVC document camera / visualiser	ABERTAY-BELLST-3014
PC	PC available for data sharing	ABERTAY-BELLST-3014
SEATING_CAPACITY	Maximum 12	ABERTAY-BELLST-3014
SLIDE_CONVERTOR	35mm slide to video convertor - Elmo	ABERTAY-BELLST-3014
VTR	JVC SVHS VTR	ABERTAY-BELLST-3014

Equipment such as the Multipoint Control Unit, gateways, gatekeepers and the bandwidth broker that have been deployed should be registered. This will make it possible for these equipment to send and receive calls.

6.2 Bandwidth broker/Gatekeeper

These two serve as a call set-up and management device which make the control of the level of H.323 IP videoconferencing traffic on the network possible. Based on the concept of “zones”, a number of venues will be under the control of a given gatekeeper. Each gatekeeper will have the different venues under its management, registered within it.

A number of bookings should be made from different registered sites. The booking system should flag off an error message if the preferred date for a booking, conflicts with dates of an existing booking. These bookings specify the desired bandwidth during that specific videoconferencing session. The booking system then seeks the confirmation of the bandwidth broker for availability of the desired bandwidth between the sites that will be engaged in such a session at the specified times.

6.3 Gateway

Gateways allow interoperation of H.323 systems with other audio/video conferencing systems on integrated services digital networks (ISDN), plain old telephone systems (POTS), asynchronous transfer mode (ATM), and other transports.

Bookings involving venues on different audio/video conferencing systems should be made to establish whether gateways that lie in this path are automatically booked. This should be the case, so that no additional booking is required.

6.4 Multipoint Control Unit (MCU)

The number of sites that can simultaneously be hosted by an MCU should be established. This, together with the bandwidth that is achievable at full capacity, should compare well with that of the product (MCU) specification, as stated by the manufacturer.

Chapter Seven

Conclusion and Recommendations

7.1 Evaluation and critical appraisal

The initial motivation for this dissertation was to engage in a piece of work that was mainly software and database oriented. Database(s) oriented because I work a lot with databases in my current job, back at home. I saw this as an opportunity to further my understanding of database concepts, which I had acquired during the course of my M.Sc. and/or current job. I wanted to take my love for Java, as a programming language, to another level – implement this task in Java so as to appreciate the power of Java as a language of choice for Internet programming and also appreciate the whole area of software engineering.

And thus, after consultations with my Supervisor, the area of a booking service for Internet – based videoconferencing was chosen. Much as I could easily identify database concepts with a booking service, the whole area of booking videoconferences over the Internet was just abstract to me. This state of affairs, therefore, meant that I spend some time understanding videoconferencing in general and Internet – based videoconferencing, in particular. There were times when I lost focus and put more efforts into understanding videoconferencing at the expense of the core area, The Booking Service.

The World Wide Web formed my major source of information. This, in itself, was a limitation in areas where I needed to physically get hold of a device under evaluation and test its performance or see how it blends (compatibility) with the other devices. The initial idea was to come up with a proto-type for the booking service. This, however, was scaled down to defining a test strategy that others should adopt if they were to test this system.

This piece of work has stretched beyond the original scheduled time and taken abnormally long. It was possible, though, to have finished this work in the original scheduled time (July and August of 2001) but various issues at different stages were at play to see me finish after this long.

At the end of the day, at a personal level, I have not achieved my objective of taking my love for Java to another level. At no point during this task did I program or use Java. This is one area I would stick to if I had to do this task (or any other, where programming is feasible) all over again.

One other area that I would have loved to spend more time on is that of **testing and validation**. Here, I mean testing and validation in a hands-on manner, as opposed to just prescribing a test strategy spelling out the areas and things to test. Thus, to a certain extent, I have lost out on the wealth of knowledge I would have gained if, say I had programmed and had to vigorously test my piece of application. One, I would gain an understanding of possible areas to test, that give one the greatest mileage and two, bugs (and the general logic and syntax) in my program would be discovered in the process.

7.2 Recommendation

The new booking service will consist of centralized equipment meant to facilitate videoconferences between sites/venues using different networking protocols and call speeds. These include the MCU, bandwidth broker, gateway, and gatekeeper equipment. IP to ISDN gateways will be deployed to provide translation services between H.320 and H.323 networks to relay audio and video streams between circuit-switched ISDN and IP networks. This will preserve legacy ISDN equipment and also accommodate any likely growth in the use of ISDN videoconferencing.

The Bandwidth Broker will provide an essential element of management, security and QoS requirements for IP videoconferencing. As mentioned before, some of the functions offered by the BB may be obtained from other nodes implementing them (e.g. a Gatekeeper), rather than these functions being implemented in the BB itself.