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DEVELOPMENT OF A DIGITAL LECTERN FOR TELETEACHING

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ABSTRACT

Increased capacity of the network will also provide the possibility to transmit pictures and sound. This is of particular interest if speeches, presentations or lectures have to be recorded or transmitted in order to make them accessible at a different location or to a larger auditorium. Very often additional information has to be transmitted simultaneously as for example the visualization of overhead slides or PowerPoint presentations with additional annotations. Up to now this has not been possible or only with a large number of staff and financial expenditure and thus is refused in many cases. This paper describes the construction and the functionality of a digital lectern named "Speaker's Corner" that solves the problems described above and also allows the use of conventional media.

Keywords: teleteaching, digital lectern, video capturing, presentation, digital annotations

INTRODUCTION

The available working speed of the computers as well as the continuously increasing transmission capacity of the network allows retrieving extensive information from the Internet. Thus it is possible to provide pictures and sound as well as video films next to pure text-based information, which leads to an increase of the number of video sequences as a carrier of information over the Internet. It is therefore obvious to use these possibilities also for education and teaching purposes. However, education makes very specific demands in the technical support required. It is characterized by a large number of necessary media that are frequently changed or used simultaneously. It is also characteristic that very often

analogous and digital media are mixed. If a new device is set up to support education and in particular lectures all requirements have to be fulfilled by the new technical possibilities.

However, in spite of these possibilities it is still very complex or almost impossible to provide sound, pictures and additional information like a PowerPoint presentation to the user over the Internet simultaneously. On the other hand there is an increasing demand for work on given project tasks without being physically present at the work place. Teleteaching, teleworking or network-based work are activities which are becoming more and more popular. They are used if the available technique is sufficient to process the information in an acceptable way. However, especially in the field of teleteaching, teleconferencing or telelearning the given technical constraints prevent the user from applying these methods in an effective way. The necessary expenditure does not justify the additional benefit. The arising costs are currently not justifiable, especially because the devices can only be used for a specific purpose.

MOTIVATION

Teleconferencing, teleteaching and telelearning have the highest technical demands on the available technology since several pieces of information must be transmitted synchronously. Especially in the field of teleteaching the available video conferencing systems or other whiteboard systems are not sufficient since they only provide a peer-to-peer connection. In many cases these systems only transfer sound and pictures to another location, additional visualizations like photographs, demonstration objects or PowerPoint

presentations were captured by the video camera instead of the speaker's image. In many cases the resolution is very poor and thus a noticeable loss of information occurs. In order to counter these circumstances the image of the speaker is not transferred but only the PowerPoint presentation with acoustic explanations. Video conferencing systems usually include no streaming facilities and thus a digital storing of the signals on a server is not possible. Due to the peer-to-peer-characteristic the number of subscribers is often limited to one station which needs specialized and expensive hardware to receive and decode the signal. However, in the field of education in particular the transmission of the audio signal and the picture of the speaker is not sufficient. The explanation of complex material requires additional types of information such as the representation of demonstration objects or the creation of drawings in addition to the prepared documents (PowerPoint slides).

In order to create a complete lecture including the speaker's image, sound and a PowerPoint presentation with additional annotations and to provide it on the Internet a large technical expenditure is necessary [1]. The speaker himself must not be impeded during the presentation so that a second person is needed to handle the technical equipment. This person must be present during the complete presentation time. Particularly the reworking of the generated data is very time-consuming since the video image of the speaker must be cut and synchronized with the PowerPoint presentation. This synchronization of the video picture with the PowerPoint presentation can only be done manually, an additional synchronization of the annotations to the PowerPoint slides has not been possible so far. For these additional tasks time is needed that is ten times as long as the actual lecture time and thus such a presentation becomes very time consuming and cost intensive. Furthermore, the extensive post-editing has the disadvantage that a large offset arises between the time of the actual presentation and its digital availability on the webserver. This chronological misalignment is not acceptable in many cases, for instance if the presentations are given in a rapid, chronological order and the listeners want to assist live or over the Internet respectively. Such a situation can occur for example within a lecture series for the repetition of the given presentation or for the preparation of the following presentation (lecture).

Up to now the problem of the synchronic integration of additional annotations in a PowerPoint presentation has not been solved. PowerPoint also offers the functionality of annotations. However, these annotations are not permanently on the slides. A paging within the slides always results in the disappearance of the created annotations. Furthermore, no ways have been found to visualize and integrate demonstration objects in a presentation. It might be necessary to employ an additional second person to operate the camera and register such objects. However, this results in a disturbance of the lecture, both for the speakers and for the listeners that assist the presentation.

CONTRIBUTIONS

In the proposed paper a possibility is introduced that allows to carry out a presentation, to acquire sound and pictures, to digitize them and to make them available to the user without a time delay. It is possible to present additional annotations to the PowerPoint slides and to integrate the visualization of transparency films or demonstration objects and to synchronize them with the presentation. Most of the tasks run automatically or can be done by the speaker without additional effort. The second person which has been needed up to now as well as the extensive reworking of the data can be eliminated and the publication time is reduced. Since the lecture is streamed and made available on a server the number of subscribers is - in contrast to a video conferencing system - almost unlimited. The recipient does not need extensive hardware anymore but free viewer programs are sufficient. However, it has to be noted that this is not a full duplex communication line but only a distributing system (one-way communication line) as it is usually needed for teleteaching. A dialogue between the speaker and the audience in the net is thus not possible.

Furthermore it is introduced how the possibilities described above are integrated in a complete device, the so-called "Speaker's Corner". The device allows the presenter to fulfill all tasks without disturbing him. With the exception of an external beamer "Speaker's Corner" contains all the necessary devices. Beside the easy operation the technical preparation time for a lecture can also be reduced to a minimum. First field reports of the use of the digital lectern in a lecture series show how the requirements were met.

REQUIREMENTS

In order to receive a comprehensive overview of the needed requirements an Internet-questionnaire has to be filled in by the potential customers, followed by a statistical evaluation which shows the most needed functionalities and requirements. The digital lectern has to fulfill the following technical requirements:

- Video capturing of the speaker: For the complete recording of all relevant information it is necessary that the speaker can be seen in the picture. It is important that no dangerous cables are placed within the auditory in order to connect the camera to the digital lectern.
- Audio capturing of the speaker: Next to the picture an important carrier of information is the audio channel. Thus it is necessary to have an audio channel that is synchronized to the picture and that is also digitized. Furthermore it has to be possible to capture the audio signal of the speaker either from an integrated microphone, to amplify and to replay it or to use the audio equipment available in the lecture room and to connect it to the digital lectern.

- Technical support of the on-site presentation: The new device is supposed to support also an on site presentation next to the task of the presentation streaming. There is for example the possibility to connect additional visualization devices such as a beamer so that the selected signal sources can be visualized. In the same way a sound system must be integrated in the device.
- Flexibility in the information delivery: An important requirement is that the system can process information from several sources. First of all it must be possible to retrieve information to be presented in advance or in real-time over a network and to copy it on the computer. Furthermore it must be possible to read data from different media as for example from floppy disk, DVD, CD or video. There should be a possibility to connect an external computer to the device. Besides this further peripherals are supposed to be connectable via an USB-connection.
For universal use the system should facilitate the use of conventional media such as paper presentations or overhead slides. It should also allow visualization of three-dimensional demonstration objects.
- Possibility for annotations: Both with conventional media as paper or transparency films and in the case of digital PowerPoint presentations the possibility to enter additional annotations must exist. These annotations must also be included in the digital recording of the presentation and should be permanently available at least for the period of the presentation.
- Possibility of online-digitalization and synchronization: All actions at the digital lectern should be registered and digitized automatically. The typical post-editing for cutting and synchronization that follows a presentation should be completely eliminated. It should be possible to provide the presentation almost synchronically on the Internet.
- Network access: In order to forward the streamed signals but also for the distribution of information a network access must be available. Likewise an additional laptop must be able to communicate with the device via a network.
- Design: The new digital lectern should have an appealing and somehow functional design. The handling of the device should be very easy also for the inexperienced user. Furthermore the functional design should allow the presenter to do all the necessary handling of the device also during the presentation. Since the device will be used in conference halls as well as in lecture halls there should be a protection against abuse or demolition.

- Intuitive usage: Usage of the digital lectern should be simple and intuitive. With that a short training period should be possible. Only the speaker is responsible for the handling of the device. His presentation should not be disturbed by this additional task.
- Mobility of the device: Due to the currently high costs of the device schools, universities and large conference hotels cannot afford a digital lectern for every lecture room. Therefore the devices must be designed to be transportable in order to move them from one room to the other.

REALIZATION

The realization of this given requirements has lead to a new product named „Speaker’s Corner“. It is described in two parts, the realization of the design and the technical realization. It is, of course the goal to optimally combine the requirements of design and technique in one single device.

REALIZATION OF THE DESIGN

For the design basically functional aspects were taken into account. However, "Speaker’s Corner" should also have an appealing outfit. At first different design studies were carried out in order to determine the best solution (Figure 1) [2].

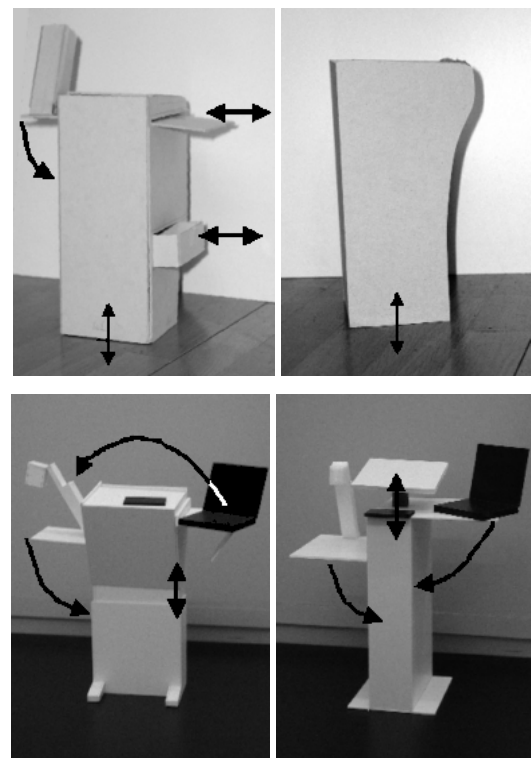


Fig. 1 Different design studies

All solutions provide sufficient space for the feet in order to allow comfortable working. A height adjustment guarantees an ergonomic height of digital lectern for the individual user. All controls for the operation of the presentation as well as for switching to the different signal sources are within the user's view and reach. This is particularly important for the WACOM tablet since this is used for the presentation and also for the annotations. Additional compartments allow the storage of additional accessories within the device. The basis of the device was designed large enough to guarantee secure handling.

The design propositions 1 to 3 in particular offer the possibility to integrate the entire equipment into a case in order to provide adequate protection against unauthorized use. This requirement is not completely achieved by design proposition 4 since the work surface of „Speaker's Corner“ remains almost unprotected. Nevertheless this proposition was realized for aesthetic reasons. The reason is that all the other design propositions look very awkward and are therefore a small stimulus for using the digital lectern. The protection against improper use in design proposition 4 was implemented in such a way that in case of non-use the entire device can be put into a box and locked. The box also provides space for all peripherals.

For the height adjustment in design proposition 4 only the work surface, i.e. the WACOM tablet, is changed but not the complete installation. When using the device additional work areas can be opened on the left and the right side of the lectern and additional space is available now to place a laptop computer. Corresponding connections at „Speaker's Corner“ guarantee both the network access and the power supply for these additional devices. As can be seen from the model the possibility exists to visualize demonstration objects, pictures or transparency films. Since transparency slides require a back illumination the table of the document camera was equipped with a film that can be electrically illuminated. The film can be turned on by a switch on the front panel of „Speaker's Corner“.

All additional elements can be lowered in case of non-use so that finally a slim pillar remains in which all modules are integrated and locked. It is not necessary to open the tables of the document camera and for the placement of the laptop computer in order to use „Speaker's Corner“. In this case only the slim pillar remains but not all the functionalities of the device can be used.



Fig. 2 General view of the device

TECHNICAL REALIZATION

The technical realization is shown in the following block diagram (Figure 3):

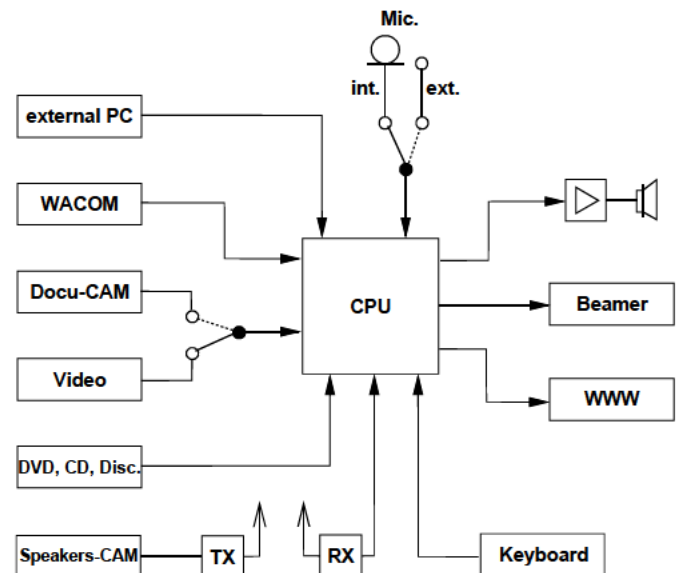


Fig. 3 Block diagram of the digital lectern

The main part of "Speaker's Corner" is a dual P4 1,7 GHz personal computer with Windows 2000 as an operating system.

All peripherals are connected to this computer, in addition it is used for the presentation of the PowerPoint slides and for the streaming of the camera signal. It provides the net accessibility and can be used to buffer the streamed data.

For the transfer of data on this computer the available network access can be used but also DVD, CD, floppy disk as well as further devices connected to the USB interface. Furthermore, an external computer can be connected to „Speaker's Corner“ via an UTP connection. In order to do this DHCP must be activated on the guest computer. This guest computer is connected via a crossed UTP cable and receives its IP-address from "Speaker's Corner" in order to realize the data exchange. The UTP-cable is firmly connected to "Speaker's Corner" and can be stowed away easily in the corpus of the lectern.

For the additional visualization of documents, transparency films or demonstration objects an additional document camera exists which can be activated by the speaker if required (Figure 4). The signal of this camera is represented both on the WACOM tablet and on the connected beamer. This allows the listeners to see the object taken by the camera and the signal is available in video stream for the remote listeners.

The head of the document camera is turnable and adjustable in its inclination. Thus it is possible to place the demonstration objects not only on the table of the document camera but also next to the digital lectern. This is of particular interest for large demonstration objects. Furthermore the document camera has a remote controlled zoom lens in order to show the whole demonstration object as well as details of it. Since the speaker is not behind the digital lectern during his explanation but next to the demonstration object the zoom lens of the document camera can also be controlled by an infrared remote control.



Fig. 4 The document camera for additional visualization

The speaker's image is taken by an external camera which is connected to the lectern" via a 2,7 GHz wireless link. Thus it is possible to transmit the camera signal without the placing of cables in the auditorium. Since the speaker is physically present in the conference room the camera signal does not appear on the beamer but is available on the streaming signal to provide the speaker's image to the external listeners. When installing a wireless link the local regulations, e.g. FCC regulations, have to be followed. These regulations might differ in different countries.

The camera is placed on the opposite wall of the lecture room, it is removed from the field of view of the audience. The camera is fix-focused onto the place of "Speaker's Corner" and thus reduces the freedom of movement of the speaker because otherwise he will leave the cameras image. However, it will be possible to have an overall view of the camera but in this case the speaker's image is too small for the remote auditorium.

Using a selector switch on the front panel of the device a video recorder or another video or audio source can be connected instead of the document camera. Depending on the position of this selector either the PowerPoint presentation, the signal of the document camera or the signal of the video recorder is active and thus visible to the auditory.



Fig. 5 Connection of the camera to "Speaker's Corner" via a wireless link

While placing the transmitter and the receiver of the wireless link of the camera it has to be ensured that both devices have a free line of sight. Objects in the propagation path of the electro-magnetic wave lead to an additional attenuation of the signal and influence the quality of the picture to be transmitted.

Therefore the listeners on site only see the picture from the selected source while the remote listener has the following content on his screen (Figure 6).

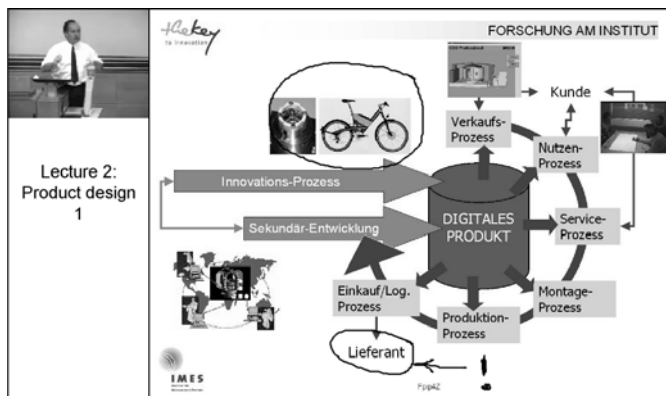


Fig. 6 Screen content for the remote user

In the upper left corner of the screen (Figure 6) is the streamed picture of the speaker. Since this picture is not the main carrier of information it was consciously chosen smaller. Tests with users in the network brought out that the speaker's image is needed for psychological reasons. Otherwise the character of a lecture gets lost. In the field below additional information can be displayed such as the title of the lecture and/or the name of the speaker. In the main window the most important carrier of information is represented, i.e. the PowerPoint presentation as well as possible annotations. If another source is selected instead of the PowerPoint presentation by the selector switch on the front panel of the device, e.g. the document camera, this signal will appear in the main window.

In addition to the video signal "Speaker's Corner" is also able to handle audio signals. An external audio signal can be

fed into "Speaker's Corner" and be amplified. A microphone can be connected to amplify the speaker's voice via the built-in audio booster. Thus the device is also usable in rooms which do not have their own audio equipment.

The central input and output device of "Speaker's Corner" is a screen with a pen-based input device instead of a mouse. The WACOM tablet PL-500 is a 15,1" flat screen that also offers the possibility to perform interactions by use of a specific pen [7]. The pen input works on an inductive principle and thus operating errors by touching the screen with the finger are impossible. This input device from WACOM allows the operation of all Windows functionalities including the Office applications like PowerPoint. The Office applications are extended by an additional software "Just Write" [8] that allows the attachment of annotations to a presentation by the use of the WACOM board (Figure 7). The annotations are also added with the inductive pen of the WACOM tablet. It is shown in Figure 7 that in the upper edge of the screen a popup-menu is available which contains additional tools for annotations like line thickness, pen color or simple geometrical figures. Furthermore, there is the possibility to add slides for further explanations. In contrast to similar Office plug-ins these annotations are lasting i.e. they do not disappear during the changeover onto the next slide. The annotations do not change the original PowerPoint file but are saved in a separate file. When restarting the presentation the speaker has to decide whether he wants to use the already available annotations or not. For large text entries or control instructions a keyboard is mounted on a telescopic slider underneath the WACOM tablet.

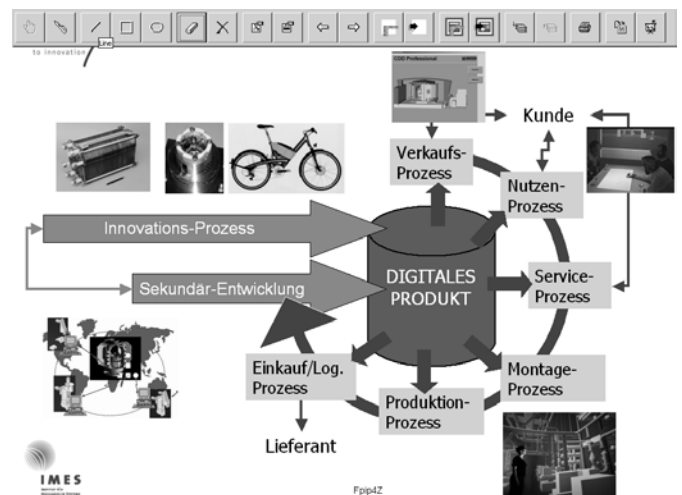


Fig. 7 Input menu "Just Write"

For the capturing, digitizing and streaming of the available signals different software packages are available. Tests showed that the software package from Real, that is Real producer, Real server and Real Player, is most suitable for the given task. The principle of the complete streaming is shown in Figure 8:

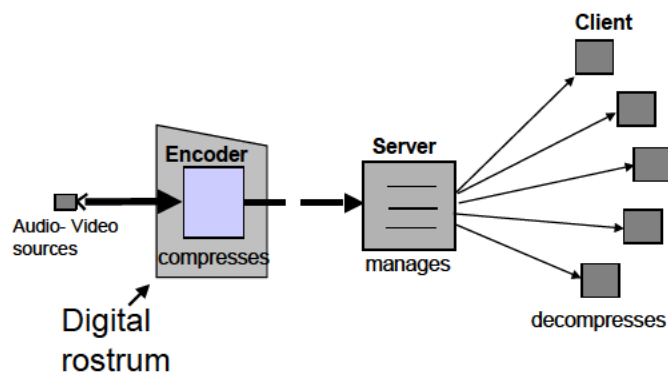


Fig. 8 The principle of streaming and the client-server architecture

The signals from the different sources are transferred to "Speaker's Corner", compressed and encoded by the Real Encoder. Afterwards the signals can be stored either locally or on a server via a network. The server manages the compressed and coded signals. He provides the signals on request to the clients with a resolution adapted to the available transmission rate. Finally the clients decode and decompress the signal and make it available to the user. If no network is available to store the data on a server, the data can be stored first locally on the hard disk of "Speaker's Corner" and written on CD after the presentation has finished.

When starting the software of "Speaker's Corner" first a software is started up which carries out the monitor recording. Then the Real Producer as well as a software "Camtasia" are started twice to record the signal from the external camera and from the monitor recording [5]. Since both processes are started almost simultaneously a synchronization between the two video streams is guaranteed. The start up of the programs is done by a batch file so that the user only has to press a single button.

In order to reduce the amount of data and to save transmission bandwidth both video streams [3] [6] have different graphical and chronological resolutions. The video picture of the speaker has a resolution of 256 x 192 pixels and a repetition rate of 15 hertz. Thus a fluent representation of the speaker, synchronized to the audio signal, is possible. The main frame has a resolution of 1024 x 768 pixels with a repetition rate of 1 Hz. Between the generation of the data and the availability on the server there is a maximum time difference of approx. 40 s which is basically caused by the compressing and the encoding. In addition non predictable delays arise from the net delay and the decompression on the client's side. The tests carried out showed that with this chosen chronological and graphical resolution a 90-minutes lecture produces a data amount of approximately 900 MBytes. For the user in the network a net transmission rate of at least 500 kB/s is required. This will be easily realized in the very near future since the capacity of the nets is increasing rapidly. Within the Intranets of schools and universities, where "Speaker's Corner" will be mainly used, this data rate is already available [4]. At lower transmission rates the amount of data must first be buffered locally on the computer.

FIRST RESULTS

"Speaker's Corner" was tested in an 8-week-experiment in education (lectures). In this experimental phase the device was in use for approximately 350 hours. The practical experience showed that the device can be productively used after a 10-minute training. Thus the training period is extremely short compared to a longer lecture series. However, it is still too long for a single lecture. With the help of a more user-friendly software the expenditure to store data on a server via a network can be reduced.

On the other hand the use of the hardware was almost free of problems. In particular the usage of the WACOM tablet was done very intuitively since the input is realized with a pen-like device. Since the appearance of the desktop (normally the PowerPoint desktop) only has insignificant changes from the software "JustWrite" no difficulties in its usage emerged. However, it turned out that the keyboard is needed during the presentation in order to flip among the slides (realized by a known PowerPoint functionality). During the presentation the pen is in the annotation mode and used for drawing and writing. Thus the pen is not in the mouse mode that is needed to flip the slides. It would be possible to change the mode within the selection menu of "JustWrite" but it is simpler to use the available keyboard instead. The handling of the digital lectern did not cause additional work for the speakers and no negative influence on the presentation was noticed.

During the test phase "Speaker's Corner" was only used in an allocated room. The device could be moved around easily within the room on the Teflon gliders. However, the gliders are not suitable for longer distances between different rooms or on an uneven floor. In order to attain better portability constructive changes and improvements have to be made. In particular the device should be transportable together with the box.

When using a wireless signal transmission of images and sound of the remote speaker's camera to "Speaker's Corner" an interference with wireless LANs installed in the same room could occur. Especially when several computers are logged in into the wireless network the channel separation between the radio-controlled LAN and the camera link was insufficient.

CONCLUSIONS

The feasibility of a digital lectern was introduced that supports a presentation consisting of PowerPoint slides, additional annotations as well as audio and video signals of the speaker. The presentation consists of different video signals which can be streamed simultaneously. Furthermore there is the possibility to use conventional transparency films and to visualize demonstration objects and to include them in the digitized video streams. A possibility is introduced to make a complete presentation available to an auditorium over a network as a live video stream or a digital record. The digital lectern was designed in such a way that the handling can be done by the speaker himself without additional operating expenditure.

The complete device was tested in lectures in order to check its efficiency and its operability. It turned out that the usage of the device is very fast and simple. However, the operator control for administrative operations can be designed in an even more user-friendly way.

In conclusion all requirements made in chapter "Requirements" were met very well by the first constructed preproduction model. So far the demand for the portability of the device has not been completely fulfilled. Since the first preproduction model was used in a single room, transportation appliances had not yet been considered.

FUTURE WORK

Future work will be on a further improvement of the operability of the digital lectern. Further tests will be carried out in order to find additional requirements and to integrate a re-worked concept into a new design.

While the present digital lectern was designed for stationary use in a lecture hall the new re-worked concept will guarantee better portability and thus a more universal use of the device. Apart from the corresponding transportation appliances the weight of the entire device has to be reduced.

A further improvement of performance of the hardware as well as a further optimization of the software (in cooperation with the involved companies) will lead to a higher resolution and a higher frame rate in the video stream. Thus the capability of the device is prepared for the higher bandwidth of the nets that is expected for the future as well as for a higher transmission capacity. This technology makes it possible to digitize and to stream a presentation which includes additional videos within the PowerPoint presentation. Further work will also deal with the compression procedures and how they can reduce the currently high amount of data. Furthermore it has to be examined which radio-controlled links are available and suitable to transmit the video signal of the remote camera. The interferences with the radio-controlled LANs should thus be eliminated. In a new model of "Speaker's Corner" a wireless microphone will be installed in order to guarantee more freedom of movement to the speaker. This is necessary for example whenever he wants to provide explanations of a demonstration object which is not on the table of the document camera.

Tests will be performed with a self-adjusting camera in order to realize a larger field of motion for the speaker without leaving the camera's image. In addition the remote user will see a larger image of the speaker since the camera is not set to an overall view but is zoomed onto the speaker.

The planned revision of the existing prototype will focus on a design that is suitable for industrial manufacturing.

ACKNOWLEDGMENTS

The equipment described above was designed in view of the necessity of providing access to lectures and presentations to a larger auditory and of making lectures digitally available to students preparing their exams. The design and the realization of the first prototype was made possible by the financing of the Center for Product Development at the ETH Zurich.

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ABSTRACT

Since 1990 West Virginia University has offered a distance education program leading to teaching certification and a master's degree in either severe/multiple disabilities or early intervention/early childhood special education. The courses were offered via satellite transmission, but in response to demand for these courses in areas without satellite access, the program is using webcasting technology to transmit class sessions over the Internet by means of Web simulcasts and re-broadcasts. The program has used two systems to deliver webcasts: Apple's QuickTime streaming format and RealNetwork's RealMedia streaming format. QuickTime streaming was easy to use, had high quality images, and offered excellent results with cable modem or fiber line Internet connections, but only fair results with 56K telephone modem access. RealMedia streaming provided lower quality video but allowed the stream to be adjusted to the user's Internet access, making the webcasts more accessible via telephone modems and lower bandwidth connections. The program decided to stay with RealMedia because it is more easily accessed by a wider range of equipment, even though overall video quality is not as good as with QuickTime. Hardware and software needs and costs are described for each system. (Contains 5 references and 15 resources on webcasting.) (TD)

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LIVE BROADCASTING ONLINE: INTERACTIVE TRAINING FOR RURAL SPECIAL EDUCATORS

Abstract

This paper describes how to use a desktop computer and inexpensive software plus a PC or Macintosh streaming server to deliver live interactive class sessions via video with audio streaming on the Internet. Although the use of Web-based instruction for preservice and inservice program delivery in special education and disability services is expanding rapidly, most existing programs rely primarily on text presentation and asynchronous (delayed time) technologies such as threaded discussions. Relatively little use has been made to date of the Web's multimedia capabilities or synchronous (real time) technologies such as audio- or videoconferencing. The use of webcasting technology (both simulcasts in real time and re-broadcasts on demand) represents a fairly inexpensive, simple to use mechanism for delivering personnel preparation programs for practitioners working in early intervention, special education, or adult disability services in rural areas without the need for high bandwidth connections. The distance education program in Severe/Multiple Disabilities and Early Intervention Special Education at West Virginia University has successfully utilized webcasting technology to deliver a graduate certification and degree program to practicing but uncertified special educators working in rural areas of the United States as well as in several other locations around the world.

Objectives

The paper will accomplish five (5) objectives:

1. to describe the application of live and archived webcasting technology to personnel preparation programs for rural special educators;
2. to illustrate hardware and software needed to create, deliver, and receive webcasts;
3. to present initial evaluation data on participant performance and perceptions;
4. to discuss the pros and cons of using webcasting for preservice and inservice training and
5. to disseminate information about resources helpful in planning for webcasting.

The procedures and outcomes described in this project may be useful for college and university faculty and/or state and local school personnel who are currently using or may be considering web-based training options in rural areas.

Introduction to the Webcasting Project

Effective and efficient personnel preparation programs at the preservice and inservice level are essential in insuring a free appropriate public education for all children, but especially in rural areas, where staff recruitment, retention, and development are critical issues. Emerging technologies offer promising solutions to the challenges colleges and universities as well as state and local education agencies face in providing accessible and appropriate training to rural special educators. The webcasting technology described in this session is an innovative approach to helping special education personnel acquire knowledge and skills needed to deliver best practices in rural schools and to acquiring full credentials for licensure/certification in special education or disability services.

Background of the Webcasting Project

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Technology-mediated distance education is rapidly becoming widely used for personnel preparation in special education at both the preservice and inservice levels. Today's telecommunications technologies allow quality programs to be delivered to many individuals effectively and efficiently without the barriers of time and space (Howard et al., 1992). A number of universities are now using distance education to address critical personnel shortages, especially in low incidence disabilities and in rural areas (Ludlow, 1998). Although web-based instruction is just now being explored for its distance education potential, it has many promising applications in teacher education in special education. Web-based instruction is especially appropriate for on-the-job training of practitioners because it allows learners to work at home or school to process materials at their own pace, to interact with the instructor and other practitioners as needed, and to receive individually designed feedback as they apply information to real world settings (Kelker et al. 1992; Kendal, 1992). Recent availability of streaming media and higher bandwidth connections now permit the delivery of live interactive sessions on the Internet in real time. Teacher education programs can harness the potential of this exciting new technology to develop more accessible, lower cost distance education programs.

Description of the Webcasting Project

West Virginia University (WVU) has offered a distance education program at the graduate level leading to teaching certification and a Master's degree in either Severe/Multiple Disabilities or Early Intervention/Early Childhood Special Education since 1990. All courses in these programs have been offered via unencrypted C-band satellite transmission to individuals attending at colleges, schools, public libraries, and other public sites throughout West Virginia and the surrounding Appalachian region. However, frequent requests by individuals living in more remote rural areas in this region, in other parts of the country without satellite access, or in international locations outside the satellite footprint (signal distribution area) prompted the program coordinator to consider how the program could accommodate these prospective students and expand its service area. In response to this demand, the program has begun to utilize webcasting technology since 2001 to transmit class sessions over the Internet to individuals without satellite access, by means of both web simulcasts and re-broadcasts.

Web simulcasts. Web simulcasts are live class sessions streamed over the Internet at the same time as satellite broadcasts. They are delivered to students who live outside the state borders of West Virginia, primarily in other areas of the United States that do not have satellite access. These students view each simulcast on a desktop computer at their home or workplace during the class session hours from 5:00 to 7:00 p.m. one night per week throughout the academic semester. They interact with the instructor and other students in the course by calling a toll-free telephone number to access the course audio conference bridging system. Both instructor and simulcast student must make allowances for a 10 to 30 second delay between the time a question is asked or a problem is posed on air and the time the distant student is able to respond to it. If the student uses a phone modem to access the Web, s/he must use a second phone line or a cellular phone to make calls during class time.

Web re-broadcasts. Web re-broadcasts are archived class sessions streamed over the Internet after each live class session that can be accessed on demand at any time. They are delivered to students who live outside continental North America where time zone differences do not permit viewing of simulcasts. They also are available to all students in the course (including those who participate by satellite transmission or simulcasts) as a mechanism for reviewing course content or as a means of making up class sessions missed due to technical problems or personal reasons. These students view each re-broadcast on a desktop computer at their home or workplace but obviously they cannot interact in real time with the instructor or other students during the live class session. To simulate live interactions with international students, the instructor provides them with discussion questions and class activities one week prior to each scheduled class session and invites them to send written comments which s/he then reads during the live class session. These students also have other opportunities to interact with their classmates through a variety of structured and unstructured online activities during the course.

Developing and Delivering Webcasts

The hardware and software needed to implement webcasting technology is readily available, simple to install and operate, and low in cost for instructor and learner. The WVU Program has used two (2) different systems

to deliver webcasts, one primarily a Macintosh computer-based system and the other a Wintel computer-based system. Each of these systems is described below along with cost estimates for setting them up.

1. QuickTime streaming format

Apple's QuickTime streaming format is the easiest system to use for webcasting, especially for those with relatively little technology expertise. The QuickTime tools are the least expensive and most user friendly for audio and video streaming, whether in real or delayed time mode (Sauer, 2001). QuickTime also has the dual advantages of cross-platform compatibility and high quality images with "skip protection" technology (Waggoner, 2001). QuickTime streaming was used for webcasts during the first year of the project, and the instructor and media producer worked with students to identify and solve problems and fine-tune the system. They concluded that Apple's QuickTime streaming provided high quality video but only a single stream of 56Kbps, which was too fast for some phone modems and too slow for high speed connections. In general, the QuickTime webcasts offered excellent results with cable modem or fiber line internet connections, but only fair results with 56K telephone modem access; the international sites were able to get good audio signal but no video signal.

The program coordinator purchased all hardware and software using internal grant funds in the summer preceding the first academic year of the project. The media producer set up an Apple OSX streaming server in a departmental office and obtained a static Internet Protocol (IP) address and a dedicated access line from university administration. He also installed Sorenson Broadcaster software on an Apple Powerbook 5300 (laptop selected for easy transport to and from the broadcast studio during classes) to create the live streams and a copy of Quicktime 4.0 Pro to convert the archived files to a Web-friendly format. Finally, he created a special icon and page within the WebCT course management system software where students could select individual links for the live and archived streams. The program coordinator wrote an extensive set of directions illustrated with screen shots to assist students in preparing their computers for webcasts; these directions were added to the WebCT course system as an HTML page with links to other Web sites.

At the broadcast studio, the production switcher was connected to the Powerbook with a cable and to another desktop computer within the studio so that the producer could monitor the live streams. The Broadcaster software was used to convert the output from the studio switcher on the fly into a Quicktime audio and video stream using the Real Time Streaming Protocol (RTSP). A single stream was sent to the program server at another location where students could access the class session via a link within the university's WebCT course management software. During the class session, the instructor monitored the simulcast with a desktop computer. After each broadcast, QuickTime Pro software was used to prepare the archived files (by means of "hinting") for delivery on demand. These archived files were then uploaded to the program server so that students could access the re-broadcasts on demand at any time throughout the course.

In this system, the instructor requires the following equipment:

- *Apple OSX streaming server equipped with a 733 MHz processor and 256 MB RAM and a 1 GB drive with a 50 stream license with no server tax (cost about \$5000)

- *Apple Macintosh G3 Powerbook 5300 with 400 MHz processor speed and 250 MB RAM to house the encoder software and facilitate transport to any site from which the class will be transmitted (for example, a broadcast studio, the instructor's office, etc.). (cost about \$3000)

- *Sorenson Broadcaster streaming software to digitize the audio and video signal at a single data rate on the fly from the camera in real time (cost about \$300)

- *XLR8 Interview external USB video capture device to convert the broadcast signal to digital video signal prior to encoding (cost about \$300)

- *Apple QuickTime Pro software (cost about \$30)

TOTAL INSTRUCTOR COST FOR QT STREAMING: approximately \$8630

This system requires students to have the following equipment:

- *desktop computer of either platform with at least 300 MHz processor and 64 MB RAM and with audio and video cards (cost varies because newer computers come equipped with such features while older computers may need to have special cards installed – since students must have computers to access other Web components of the course, this generally requires no additional costs to them)

- *free copy of Apple QuickTime Player 4.0 downloaded to their computer's desktop and installed in their browser's plug-ins folder to view the webcasts (no cost)

TOTAL STUDENT COST FOR QT STREAMING: none

2. RealMedia Streaming Format

RealNetwork's Real Media streaming format is another streaming system that is not too difficult or costly to use for webcasting and is better suited to the varying nature of bandwidth currently available to end users on the Internet. RealMedia streaming has the advantage of producing several streams at different access rates or using patented Surestream technology to offer a single stream that automatically adjusts to the user's access rate (Waggoner, 2001). For this reason, the RealMedia tools are more expensive but offer greater flexibility (Sauer, 2001). In order to insure greater access to the webcasts, the program coordinator and media producer investigated other options for transmitting live and archived video streams, deciding to use RealMedia streaming for webcasts during all subsequent years of the project. They concluded that RealMedia streaming provides lower quality video but the Surestream technology allows the stream to be adjusted to the user's Internet access, making the webcasts more accessible via telephone modems and lower bandwidth connections. The Real Media webcasts offered excellent results with cable modem or fiber line internet connections, and good results with 56K telephone modem access; the international sites were able to get good audio signal and fair video signal.

The program coordinator and media producer collaborated with technical personnel at the West Virginia Education Network (EDNET) in Institute, West Virginia to transmit live webcasts and record archived webcasts for the remaining courses in the project. EDNET personnel set up a rack-mounted Athlon ATX computer to run the encoder software. They purchased an annual 100 stream Real Producer Pro license from RealNetworks to encode the streams on the fly from the live satellite broadcasts and installed an internal PCI video capture card to convert the analog signal sent from the WVU broadcast studio to digital signal. They used a Microsoft Windows NT 4.0 server to store and serve the live and archived video streams.

In this system, the instructor requires the following equipment:

- *Microsoft Windows NT server equipped with a 700 MHz dual processor and 86 MB RAM plus a 10 GB drive (cost about \$3500)

- *RealNetworks RealProducer Pro streaming software to encode simulcasts and re-broadcasts with a 100 stream license (cost about \$2000)

- *Athlon ATX computer with 500 MHz processor and 64 MB RAM to house the encoder software (cost about \$1000)

- *Microsoft Windows 2000 Pro server software (cost about \$1000)

- ViewCast Osprey 100 internal PCI video capture card to convert the broadcast signal to digital video signal prior to encoding (cost about \$200)

Additional equipment needed if no broadcast facility is available:

TOTAL INSTRUCTOR COST FOR RM STREAMING: approximately \$7700

This system requires students to have the following equipment:

- *desktop computer of either platform with at least 300 MHz processor and 64 MB RAM and with audio and video cards (cost varies because newer computers come equipped with such features while older computers may need to have special cards installed – since students must have computers to access other Web components of the course, this generally requires no additional costs to them)

- *free copy of RealNetworks RealMedia Player 8.0 Basic downloaded to their computer's desktop and installed in their browser's plug-ins folder to view the webcasts (no cost)

TOTAL STUDENT COST FOR RM STREAMING: none

NOTE: In the WVU program, class sessions are broadcast for satellite transmission from either a television studio or an electronic classroom, so the audio and video signal was taken directly from the studio's switcher and cabled into the laptop computer for conversion to the streaming format. However, several trials were also successfully conducted in which the simulcast was streamed directly from the instructor's office using a digital camera and wireless microphone.

Additional equipment needed if no broadcast facility is available:

- *Sony DVX1000 digital camera to record video signal when not in a television production facility (cost about \$3000)

- *Sony ECM44BC omni-directional lavalier microphone to record audio signal from instructor to the camera (cost about \$200)

- *Bogen 3001 BN video tripod with fluid head (cost about \$150)

ADDITIONAL COST IF NO STUDIO ACCESS (EITHER SYSTEM): \$3350

Implications of the Webcasting Project

The WVU Webcasting Project represents an innovative model of Web-based training that has been developed, implemented, and evaluated with practicing special educators and related services specialists in rural areas, both locally and globally. The availability of web simulcasts has enabled students living in states other than West Virginia to participate in the certification and degree program; to date, 10 students have enrolled in courses from states such as California, Colorado, Florida, Indiana, New Jersey, New York, North Carolina, and Virginia. The first student to have completed the program via the simulcast model will graduate in Spring 2003. The simulcasts are also used frequently by students in West Virginia when they cannot access the satellite broadcasts due to technical problems at their site or if they must be absent due to illness or bad weather. The availability of re-broadcasts has allowed six (6) individuals living in international areas to join the program; four students have enrolled from Saipan, an island in the South Pacific, one has enrolled from Iceland, and another from Japan. The first international student to complete the program via re-broadcasts graduated in Spring 2002 and two more will finish in Spring 2003. The re-broadcasts are also used often by all students to review content or activities from the broadcast/webcast sessions as needed throughout the course. Beginning with the Spring 2002 semester, program staff no longer make videotape copies of class sessions for students who miss class because they can access the webcasts online more quickly and with less effort. The program coordinator has made the decision to stay with the

Real Media video streaming for the time being, because it is more easily accessed by a wider range of equipment, even though the overall quality of the video image is not as good as the Quicktime streaming.

Webcasting is now a permanent component of the distance education program in Severe/Multiple Disabilities at WVU and it may also be appropriate for other personnel preparation programs in rural areas. The advantages of webcasting include enhanced access to training programs in even the most remote rural areas; reduction in the cost and effort associated with duplicating videotapes copies of class sessions; and provision of opportunity for national and even international outreach. The disadvantages of webcasting are the lower quality of the video image online compared with high quality images via satellite; the bandwidth limitations in some areas of the country or the world; and the continuing technical difficulties with the streaming server and the users' computers. Nevertheless, webcasting represents a viable option for delivering instruction in real and/or delayed time as well as an acceptable balance between program quality and accessibility.

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Resources for More Information about Webcasting

Apple products are available from <http://www.apple.com>
Sorenson Broadcaster is available from <http://www.sorenson.com>
XLR8 products are available from <http://www.xlr8.com>
Microsoft products are available from <http://www.microsoft.com>
Real Networks products are available from <http://www.real.com>
Athlon products are available from <http://amd.com>
Viewcast products are available from <http://viewcast.com>
Sony products are available from <http://www.sony.com>
Bogen products are available from <http://www.bogen.com>
Discount prices on media products are available from <http://bhphoto.com>

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Digital Video, a magazine available at no cost from <http://www.dv.com>

AV Multimedia Producer, a magazine available at no cost from <http://www.avvideo.com>

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Original Paper

Bandwidth Constraints to Using Video and Other Rich Media in Behavior Change Websites

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Abstract

Background: Web-based behavior change interventions often include rich media (eg, video, audio, and large graphics). The rationale for using rich media includes the need to reach users who are not inclined or able to use text-based website content, encouragement of program engagement, and following the precedent set by news and sports websites.

Objectives: We describe the development of a bandwidth usage index, which seeks to provide a practical method to gauge the extent to which websites can successfully be used within different Internet access scenarios (eg, dial-up and broadband).

Methods: We conducted three studies to measure bandwidth consumption. In Study 1, we measured the bandwidth usage index for three video-rich websites (for smoking cessation, for caregivers, and for improving eldercare by family members). We then estimated the number of concurrent users that could be accommodated by each website under various Internet access scenarios. In Study 2, we sought to validate our estimated threshold number of concurrent users by testing the video-rich smoking cessation website with different numbers of concurrent users. In Study 3, we calculated the bandwidth usage index and threshold number of concurrent users for three versions of the smoking cessation website: the video-rich version (tested in Study 1), an audio-rich version, and a Web-enabled CD-ROM version in which all media-rich content was placed on a CD-ROM on the client computer.

Results: In Study 1, we found that the bandwidth usage index of the video-rich websites ranged from 144 Kbps to 93 Kbps. These results indicated that dial-up modem users would not achieve a “good user experience” with any of the three rich media websites. Results for Study 2 confirmed that usability was compromised when the estimated threshold number of concurrent users was exceeded. Results for Study 3 indicated that changing a website from video- to audio-rich content reduced the bandwidth requirement by almost 50%, but it remained too large to allow satisfactory use in dial-up modem scenarios. The Web-enabled CD-ROM reduced bandwidth requirements such that even a dial-up modem user could have a good user experience with the rich media content.

Conclusions: We conclude that the bandwidth usage index represents a practical tool that can help developers and researchers to measure the bandwidth requirements of their websites as well as to evaluate the feasibility of certain website designs in terms of specific use cases. These findings are discussed in terms of reaching different groups of users as well as accommodating the intended number of concurrent users. We also discuss the promising option of using Web-enabled CD-ROMs to deliver rich media content to users with dial-up Internet access. We introduce a number of researchable themes for improving our ability to develop Web-based behavior change interventions that can better deliver what they promise.

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KEYWORDS

Health behavior; Internet; behavioral research; rich media; video; smoking cessation; bandwidth usage index

Introduction

The Internet holds great promise as a delivery channel for programs designed to help people change their behaviors [1,2]. For example, a number of reports have described encouraging results for Web-based interventions for quitting smoking [3-7], managing diabetes [8], managing depression and stress [9,10], and increasing exercise and losing weight [11,12]. An early review of this burgeoning field concluded that Web-based interventions were relatively more helpful than non-Web controls across a broad range of behaviors and methodologies [13]. One of the more impressive features of Web-based behavior change programs is their ability to incorporate rich media components that use video and audio.

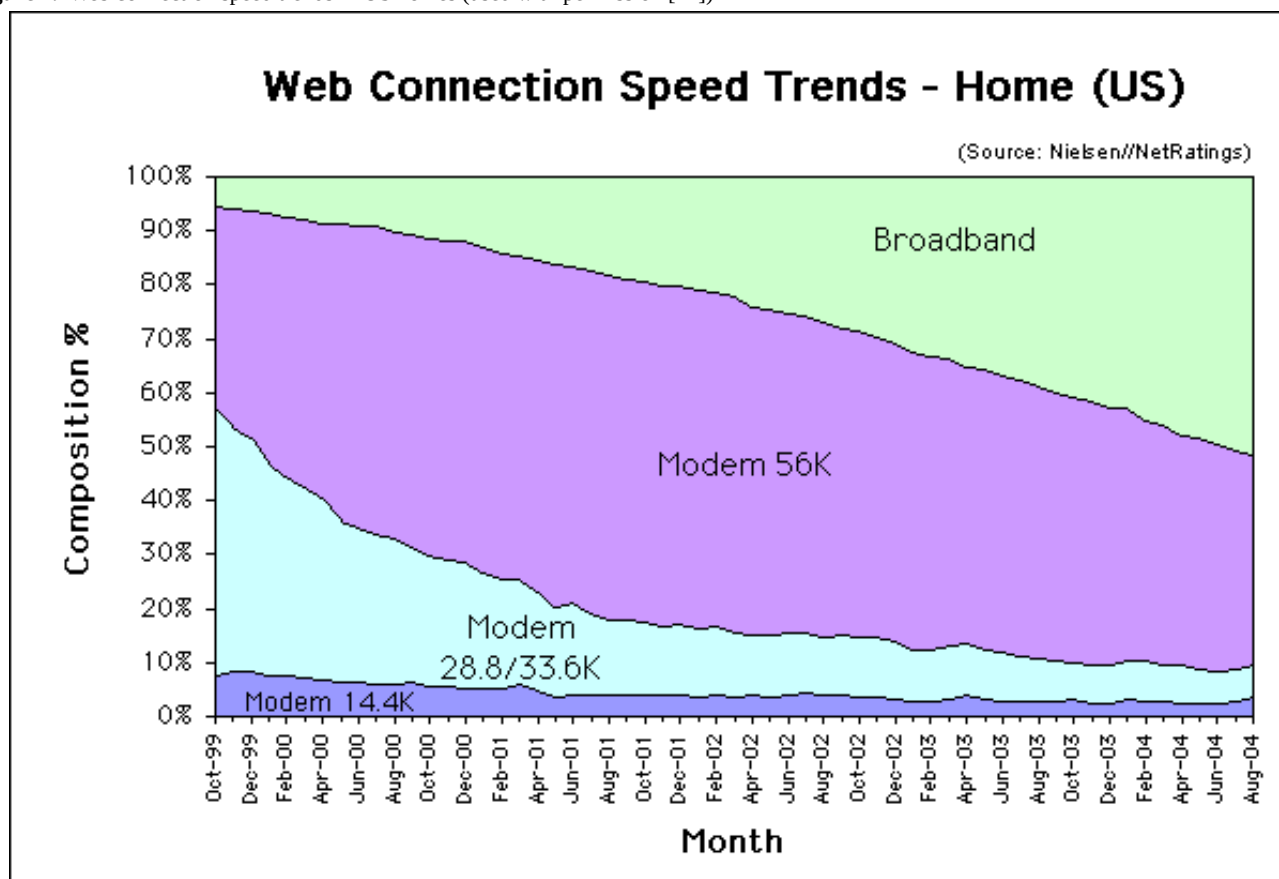
In this report we will review some of the forces that encourage the development of rich media websites, but we will also examine in considerable detail the factors that temper unbridled enthusiasm for this trend. We believe that any reasoned analysis of the use of rich media website content should also consider the possible barriers to its use. As a key part of our analysis, we will describe the development and testing of a practical index for gauging the amount of bandwidth that a Web-based program requires. We hope to highlight the ways that bandwidth constraints associated with rich media restrict both program reach as well as the number of concurrent end users who can successfully use a Web-based program. We also discuss ways that behavioral research can help elucidate how and when to

use different website program ingredients, including, for example, the use of rich media. The remainder of this section describes the trends encouraging the use of rich media and introduces key technology issues that set the stage for the presentation of our investigation.

Trends and Assumptions That Encourage Use of Rich Media

Developing rich media is an expensive proposition [14]. Nonetheless, the use of rich media in Web-based programs has been encouraged by complementary technology trends: (1) the dramatic reduction in the cost and complexity of rich media recording and editing tools, (2) the fact that rich media is a defining characteristic of CD-ROM multimedia training [15-17], (3) the emergence of programming tools that reduce the complexity of delivering audio and video in websites, (4) the use of rich media content in a number of commercial websites (eg, sports, news, and entertainment sites) and company intranets [18], and (5) the growing number of households with *broadband* Internet access both in the United States [19] and the world [20]. Currently, more than 204 million people in the United States have home Internet access [19]. In August 2004, 51.4% of active Internet users connected from home using broadband connections. Among *narrowband* users, 39% use 56 Kbps modems, 6% use 28/33.3 Kbps modems, and 3.6% use 14.4 Kbps modems [19,21]. In summary, just under half of home users in the United States connect to the Internet at 56 Kbps or less (Figure 1).

Figure 1. Web connection speed trends in US homes (used with permission [21])



The emergence of rich media websites has also been encouraged by a number of assumptions made by behavior change program developers: (1) online video and audio presentations help reach the audience that is not able or inclined to interact with websites that use text-based content; (2) rich media may be more effective than text in conveying emotional content and subtle interpersonal communications [22]—issues that may be particularly pertinent to many behavior change programs; (3) by varying rich media components, Web-based programs can deliver tailored content that is more closely aligned with the racial, ethnic, and demographic characteristics of users (eg, varying the ethnicity or age of video models can affect acceptability and impact [23]); and (4) the entertainment value of rich media can help overcome problems of participant recruitment and program engagement that seem to afflict many Web-based programs. A recent study on cancer communication materials found that participants preferred multimedia which combined text, spoken audio, and animation compared with presentations that involved text only, audio only, or text and synchronized audio [24,25].

Technical Considerations

In this section we briefly introduce a number of technical considerations that affect website bandwidth constraints, including the size of rich media, ways to deliver rich media, calculating the extent of concurrent website usage, and typical bandwidth bottlenecks. Our discussion will only briefly touch upon the vast wealth of technological detail and resources associated with Internet content delivery/distribution. We encourage interested readers to explore this realm by first consulting the useful framework presented by Bush and his colleagues [26], followed by review of several excellent texts [27-29].

Web-Delivered Rich Media

Rich media content delivered over the Web typically undergoes a number of changes in order to reduce its absolute size (number of bits of data) while maintaining an acceptable level of clarity and attractiveness (fidelity to the original) [30-32]. The smaller the data size of a media file, the more rapidly it can be delivered over the Internet to the end user. One approach to reducing the data size of rich media involves scaling, that is, reducing the frame or image size, the frame rate, or the color resolution [27]. In addition to scaling, data size can be reduced by using very sophisticated technologies that compress (remove information that is perceptually redundant [27]) media content with varying degrees of fidelity and acceptability [28]. Video can be delivered using a *progressive download* process or via *streaming*. In our experience, progressive download provides better fidelity with fewer interruptions and fewer problems synchronizing video with audible speech (correctly timing speaker's lips and voice). Synchronization problems of this type are very distracting, and, as a result, they significantly degrade usability [33]. Finally, it is possible for websites to *preload* or *push* rich media to the end user before such content is selected for viewing [34], but this approach works best when the user has a broadband connection and when rich media selections are few in number or can be predicted reliably ahead of user selection (not typical of rich

media behavior change websites). Corporate examples of preloading can be found in media content stored in cache on proxy servers [27].

Concurrent Users

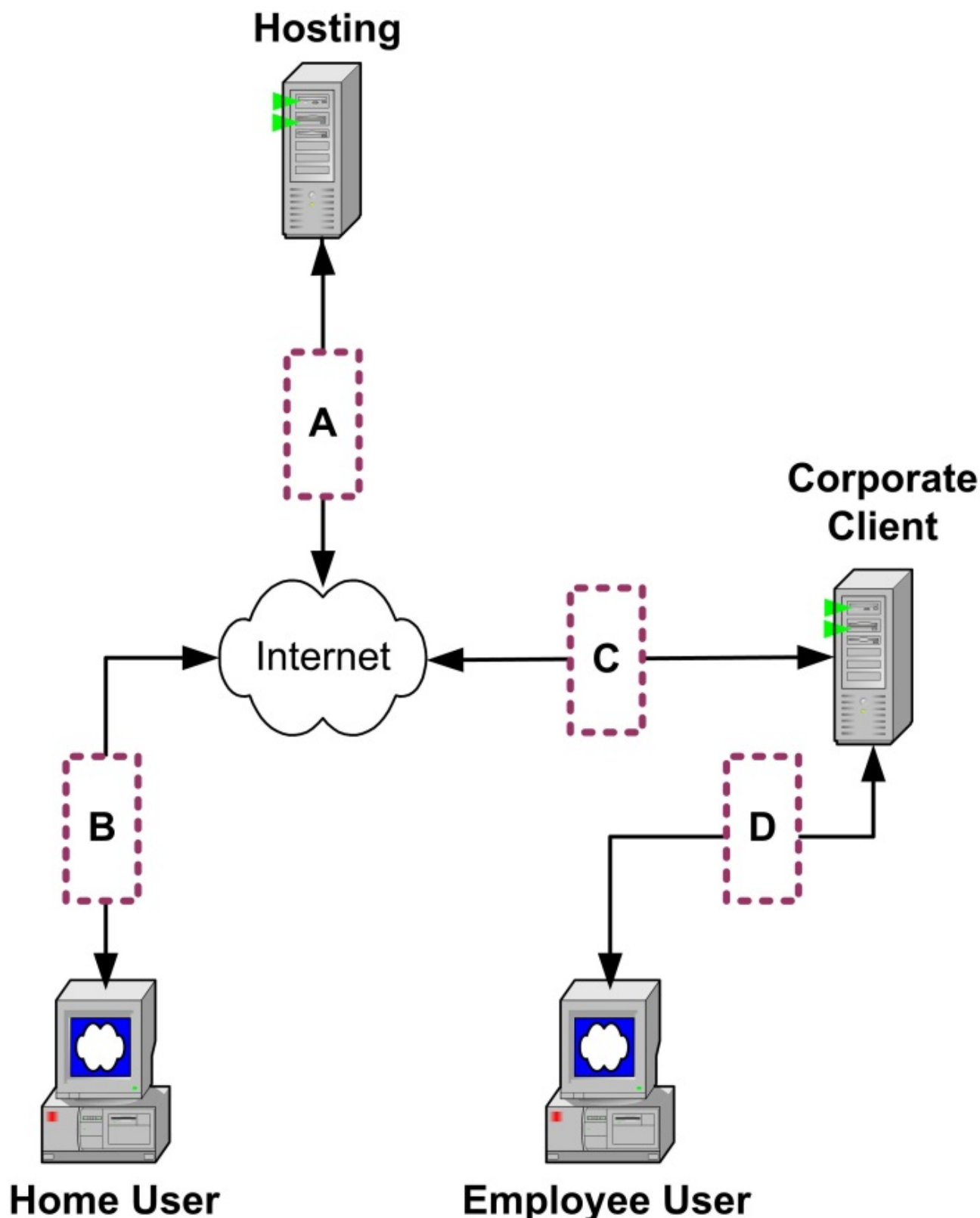
Website usability is greatly affected by the number of users who access content at the same time. As a general rule, because of the asynchronous use pattern associated with most Web-based programs, the number of concurrent users is a small percentage of the total number of users eligible to access the program. However, given that websites can be made available to and thus accessed by an extremely large number of users, even a small percentage can yield a relatively large absolute number of concurrent users. As Salchner [35] has noted, concurrent user estimates can be derived by calculating the average number of user visits that occur each day and the average number of users per hour that a program can support (60 divided by the duration [in minutes] of a typical session or visit). Of course, visits are not evenly distributed across the hours of the day. And each user does not access the same program content (especially in Web-based programs with matrix information architecture components that enable users to choose how they access program content [2]). In summary, the number of concurrent users is critically important to website usability since it interacts with the available bandwidth (described next).

Bandwidth Bottlenecks

Most users and program developers are familiar with the term *bandwidth*, which is defined as “the amount of data that can be transferred through a digital connection in a given time period...[which is] usually measured in bits or bytes per second” [36]. The physical metaphor of a pipe is often invoked in an effort to describe the manner in which digital connections behave: larger pipes have greater bandwidth and they permit faster throughput of the digital data. As with the limits imposed by physical pipes, there are limitations on the amount of data that can be transmitted through any digital connection.

Bandwidth constraints have significant practical implications because, when exceeded, they can greatly increase end user frustration and seriously degrade interaction with a website. Consider, for example, that usability research confirms that users have extremely limited patience when waiting for a Web page to download [37]. The longer the wait, the less acceptable—and hence the less usable—the website. Jakob Nielsen, a noted Web usability expert, has recommended that response times of less than 0.1 second are needed for the user to feel that the system is being responsive (that the system has “heard” and “responded” to the user's request) and that any delay of greater than 1.0 second will interrupt the user's flow of thought [38].

There are at least four potential delivery bottlenecks that can interact in a cumulative fashion to impede the acceptable flow of content from the Internet to the end user [39]. The size/speed of the digital connection at various key points in the content delivery process has a very significant impact on bandwidth. Figure 2 depicts the key points where throughput bottlenecks can occur.

Figure 2. Potential bandwidth bottlenecks

Bottleneck A is defined as the capacity of the host to serve (deliver) the program content. Throughput can be undermined at this point if the program content is being served by an inefficient or overworked server. Fortunately, current servers are sufficiently robust that they become a throughput bottleneck only in scenarios when they are called upon to manage responses and generate logic to build and delivery dynamic Web pages

for a very large number of concurrent users. Moreover, there are sophisticated hardware and software solutions for handling this type of demand. For example, major commercial Internet content providers typically use a server farm configuration in which one group of servers manages applications (program logic), other servers process database activities, and yet other servers manage the delivery of media assets (eg, graphics, audio,

and video) [40]. However, when many concurrent users access video-rich content, the bottleneck is defined by the bandwidth constraints that cannot be resolved by improvements in server hardware configuration.

Bottleneck B describes the connection between the Internet and the home user. It is possible that the user's computer can be underpowered to manage Web-based programs with rich media. But a critically important factor at this point involves the "last mile" challenge [41,42] (ie, bandwidth limitations in delivering Internet content from the service provider to the home user). The limited bandwidth of dial-up modem users represents an obvious example of this bottleneck. So-called *broadband* connections enable home users to download significantly greater amounts of data per unit time from the Internet.

Bottleneck C describes the connection between the Internet and a company network. Many companies use T1 lines that provide all employees with "high speed" access to the Internet, email, and data sharing.

Bottleneck D describes the connection between a company network and the individual employee workstation.

Summary

Rich media is becoming a frequent ingredient of Web-based behavior change programs. As we have noted, there are forces that encourage this trend, as well as associated bandwidth requirements that need to be measured in order to inform the

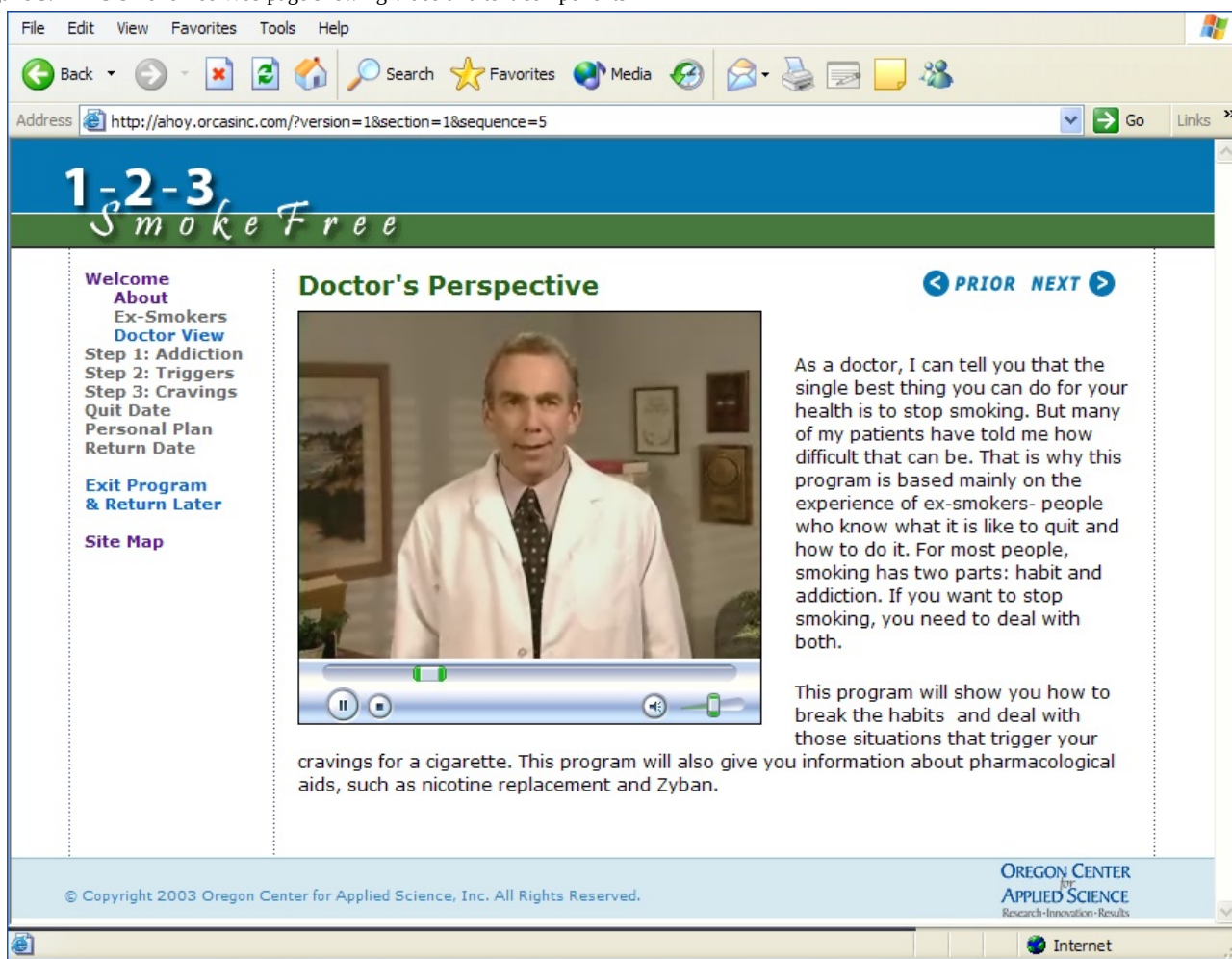
optimal planning, development, and delivery of Web-based programs that use rich media.

Methods

Study 1: Three Different Web Programs

We used a relatively simple methodology to calculate the bandwidth requirements of an Internet-based program. The resulting metric, which we have labeled the bandwidth usage index (BUI), can be used to define the minimum bandwidth requirements for any given program and can be used to estimate the number of concurrent users that can be accommodated (have a good user experience) under varying Internet access scenarios (eg, ranging from using a dial-up modem to using an internal network).

We determined the BUI for three different video-rich Web-based behavior change programs: (1) the 1-2-3 SmokeFree smoking cessation program [43] (Figure 3), (2) a program designed for caregivers, and (3) a program designed to improve eldercare by family members. Each of these programs was developed with NIH Small Business Innovation Research funding at the Oregon Center for Applied Science, each used a modularized, data-driven design that displayed content based upon the interaction of logic scripts (eg, PHP), SQL databases, and cascading style sheets, and each made extensive use of video and audio (rich media) components. In all tests, video was delivered using the progressive download method via a Windows Media Player object embedded in the Web page.

Figure 3. 1-2-3 SmokeFree Web page showing video and text components

Specifically, we measured the amount of data received by an end user (client computer) from a server (host computer) that delivered representative content from each of the three rich media Web-based programs. The 13-minute test followed a consistent protocol described generically in [Table 1](#) and in more

technical detail (with an example) in [Multimedia Appendix 1](#). All tests were conducted in a 100 Mbps Ethernet network, which essentially eliminated any meaningful artifacts that would be associated with delays in network transmission (server response and bandwidth constraints).

Table 1. BUI measurement protocol

Step	BUI Measurement Task
1.	Clear the browser cache (from MS Internet Explorer 5 x, select "Tools," Internet Options," "General" tab, and click on "Delete Files" button).
2.	Access statistics for the network port that will be used by the test computer.
3.	Determine the noise level during a period of inactivity (see Multimedia Appendix 1).
4.	Reset the byte counters or settings for the port connected to the test computer.
5.	Launch and then use the Web-based behavior change program in a representative manner for the designated test period.
6.	Calculate the bytes received by the test computer at the end of the test period.
7.	Subtract the predicted noise from the bytes received calculation.
8.	Convert (as needed) into Kbps to obtain the BUI.

It is important to note that the end user in this test accessed a representative portion of the behavior change program content hosted on the server. We defined "representative portion" as a mixture of Web pages which included text, graphics, and rich media content (audio and video) used in a manner that simulated

a typical use case. The index or gauge of data throughput was calculated as the total number of bits received by the client computer from the server divided by the duration of the test period to yield an average expressed in Kbps.

We used the BUI calculations to predict the number of concurrent users who could have a satisfactory experience using Web-based behavior change programs under a number of different Internet access scenarios.

Study 2: BUI Validation Tests

In Study 2, we attempted to confirm the validity of our estimated threshold number of concurrent users for the three rich media behavior change websites we tested in Study 1. First, we used operating system tools to *throttle* a network computer (host) so that it would allow only 768 Kbps throughput (equivalent to the maximum throughput that would be experienced when using a DSL/cable to access the Internet). Next, we used that network computer to host the video-rich 1-2-3 SmokeFree Web-based smoking cessation program tested in Study 1. We stationed each of five end user participants (blind as to our earlier calculations and hypotheses) in separate offices where they could receive study instructions. Each end user participant used a separate workstation connected to the company intranet, and each was asked to clear the browser cache (Step 1 in Table 1). Finally, each participant was instructed at the same time via the intercom to launch and then use the Web-based behavior change program in a representative manner, which was described to them in terms of activities to perform and pages to visit (Step 5 in Table 1). At the end of the 5-minute test period, each participant was asked to rate the quality of their experience with the Web-based program using the binary label of “a good experience”—the program pages flow with little or no delay—or a “bad experience.”

Study 3: Varying BUI in Three Versions of the Same Web Program

We also tested the extent to which the bandwidth requirements of a single Web-based program might vary based upon the type of rich media it presents and its delivery format. More specifically, we created two new versions of the 1-2-3 SmokeFree Web-based smoking cessation program: (1) an “audio-rich” version in which program content had been adapted such that video content was replaced by audio-only content, and (2) a Web-enabled CD-ROM condition in which all media-rich content was placed on a CD-ROM that was located on the client computer. All program content and behavior was otherwise identical to that found in the video-rich version of the smoking cessation program tested in Study 1. We followed the test protocol described in Table 1 to obtain throughput measures (the BUI) for each of these new versions in order to determine how they compared to the video-rich version.

Results

Study 1: Three Different Web Programs

The BUI for three video-rich Web-based behavior change programs ranged from 144 Kbps for the smoking cessation website to 93 Kbps for the eldercare website (Table 2). By dividing the total amount of data that can be transmitted (based upon the type of Internet connection) by the BUI (the amount of data per second being transmitted by the program), we were able to estimate the number of concurrent users who could use the Web-based program content. It is important to note that we used the *idealized* throughput of various methods of accessing the Internet, ranging from the slowest 28.8 Kbps dial-up modem to an extremely fast 100-baseT corporate network [44,45], to calculate the estimated number of concurrent users who could be accommodated for each website. Many factors affect throughput availability (eg, network noise, competition from other applications or other users, and server and end user hardware and system configurations). Actual throughput is almost always less (never greater) than these idealized numbers. As a result, a more precise calculation of the maximum number of concurrent users would require the BUI (the focus of this report) and a representative measure of throughput availability.

In terms of Bottleneck A (Figure 2), Table 2 provides a useful guide for how many concurrent users an Internet host could accommodate. Internet service providers (ISPs) typically offer a range of services, including increasing levels of bandwidth at correspondingly higher costs. If the ISP offers Internet access equivalent to T1 bandwidth, then Table 2 shows that the ISP could host only 10 to 16 concurrent users of the rich media programs (eg, the smoking cessation program had $1544 \text{ Kbps} / 144 \text{ Kbps/user} = 10 \text{ users}$ [an integer]).

In terms of Bottleneck B describing end user access to the Internet, data displayed in Table 2 show that users with a dial-up modem (either 28.8 or 29 Kbps or 56 Kbps) would not be able to achieve a “good user experience” with any of the three rich media website programs. In order to use any of these programs, the end user would need to have at least a DSL/cable broadband connection.

In terms of Bottleneck C, Table 2 indicates that all Internet activity would be brought to a halt in a company with a single T1 access to the Internet if 17 or more of its employees were to concurrently access any of the rich media websites. Moreover, the same result would be obtained if only 11 employees were to concurrently access the smoking cessation program. Generally speaking, companies are loath to have mission-critical business activity interrupted in this manner.

Table 2. Estimated number of concurrent users accommodated by three behavior change Web-based programs, by type of Internet connection

Type of Internet Access and Related Maximum Download Throughput	Concurrent Users of Rich Media Web-Based Programs			
		Smoking cessation	Caregiver	Eldercare
Connection Type	Kbps	BUI = 144 Kbps	BUI = 95 Kbps	BUI = 93 Kbps
Dial-up	29	0	0	0
Dial-up	56	0	0	0
DSL/cable	384	2	4	4
DSL/cable	768	5	8	8
T1	1544	10	16	16
Home WIFI from cable modem	1972	13	20	21
Home cable modem	3000	20	31	32
T3	44736	310	470	481
Internal network (100-baseT)	100000	694	1052	1075

Finally, considering Bottleneck D, data presented in [Table 2](#) underscore the fact that the remarkable speed of network connections means that this connection would rarely become a bandwidth bottleneck. This conclusion would seem to recommend that rich media programs should be placed on a company server or, alternatively, on a server that is connected inside the company firewall. However, few corporate IT departments wax enthusiastic about outside servers being connected to their networks, and there is also the challenge of outside organizations being able to maintain and update these servers within idiosyncratic IT environments. It is also important to note that Web-based programs are typically developed and then demonstrated using corporate networks that present few bandwidth constraints. Tests on corporate networks can lull the user/client/developer into the mistaken conclusion that the program will give a good user experience when it is delivered over the Internet.

Study 2: BUI Validation Tests

When 5 participants accessed the video-rich version of the smoking cessation program using a maximum download bandwidth of 768 Kbps, all users reported that they had a good experience with the program. The situation changed dramatically when 6 participants concurrently accessed the same program from within the same network environment—in this test case, all users reported that they had an unsatisfactory experience using the website, with reports of long delays in page loading, videos not playing, and so forth. These results confirmed the validity of the threshold numbers described in [Table 2](#). Specifically, using the metrics in [Table 2](#), it is possible to predict

the number of concurrent users that will have a satisfactory experience. Exceeding that threshold number completely retrogrades the satisfactory experience for each concurrent user. The effect is not partial but, rather, it leads to a complete inability to access the program.

Study 3: Varying BUI in Three Versions of the Same Web Program

Results of the third study are presented in [Table 3](#). Note that changing a program from video to audio reduced bandwidth requirements by almost 50% (from a BUI of 144 Kbps to 65.9 Kbps), but this bandwidth requirement was still too large to enable an end user to access this audio-rich Web-based content using a dial-up modem. In marked contrast, using the Web-enabled CD-ROM reduced bandwidth requirements dramatically (from a BUI of 144 Kbps to 2.1 Kbps), which would enable a user to easily access rich media content even from a dial-up modem. It is also important to note that even if a Web-enabled CD-ROM were to be used, the hardware and software constraints become increasingly salient when Internet access speeds increase (eg, T3 and faster), thereby enabling more concurrent users to access hosted content. Consider, for example, that [Table 3](#) indicates that by using a T3 access to the Internet and a Web-enabled CD-ROM program format it might be possible to host more than 21000 concurrent users. In such circumstances, greater attention must be afforded to the number and type of server configurations (hardware and software) needed to generate logic, select content, and build dynamic Web pages.

Table 3. Estimated number of concurrent users accommodated by three variants of the same Web-based smoking cessation program, by type of Internet connection

Type of Internet Access and Related Maximum Download Throughput	Concurrent Users of Variants of a Web-Based Smoking Cessation Program			
	Video-Rich		Audio-Rich	Web-Enabled CD-ROM
	Kbps	BUI = 144 Kbps	BUI = 65.9 Kbps	BUI = 2.1 Kbps
Dial-up	29	0	0	13
Dial-up	56	0	0	26
DSL/cable	384	2	5	182
DSL/cable	768	5	11	365
T1	1544	10	23	735
Home WIFI from cable modem	1972	13	29	939
Home cable modem*	3000	20	45	1428
T3	44736	311	678	21302
Internal network (100-baseT)	100000	694	1517	47619

* Faster cable modem download speeds up to three times this figure are emerging as providers ease bandwidth restrictions.

Discussion

In this report we have described a series of tests of the bandwidth usage index (BUI) that confirm the value of measuring the actual data throughput associated with delivering Web-based behavior change programs that have varying amounts of rich media. We believe that the BUI or other similar throughput measures can provide designers with important guidance about how much rich media to include in their Web-based programs depending upon the intended audience and the intended number of concurrent users.

Diffusion of Technology Changes

While technology changes at a very rapid rate, it is nonetheless also true that program designers need to take into careful consideration the significant delays associated with the diffusion of these changes to the large installed base of computer users. This challenge is confronted when considering end user software and hardware (eg, types and versions of operating systems, browsers, and browser plug-ins like Macromedia Flash and Adobe Acrobat; typical monitor sizes; and screen resolution settings). It is also highly relevant when considering types of Internet access which, in turn, affect the choice of program ingredients. These considerations may be even more salient when a Web-based intervention is targeted to groups that are not as far ahead as others on the adoption curve—as in the limited broadband penetration in rural [46] and different ethnic and racial communities [47]. Moreover, the adoption rate for broadband Internet access may slow down [21,48].

Tradeoffs to Delivering Digital Content

Delivering digital content is fraught with tradeoffs regarding costs, reach, and scalability. If a Web-based program uses rich media content, then the calculations we describe in this report indicate that the program will not be usable for approximately half of US Internet users who still use dial-up modems. Even if you target a website to users whose broadband Internet access

provides them with more than sufficient bandwidth to receive rich media programming, the hosting of that rich media content to thousands of concurrent users has substantial costs associated with using content delivery network solutions (eg, Akamai [49] and Cisco Systems [40]). Commercial websites subsidize such outsourcing by selling paid Web page advertisements—a business model that may not be feasible or even appropriate for many behavior change websites.

Recommendations for Future Research

We recommend that reports of Web-based programs should routinely provide an indicator of bandwidth consumption, like the BUI, in order for readers to evaluate the manner in which such programs can be used. Of course, not all Web-based programs need to be designed for large audiences. But those programs that have widespread use as their goal need to be designed—and then fully tested—in a manner that ensures they can deliver the goods consistent with their intentions.

As we noted in our introduction, we believe that empirical research needs to examine the accuracy of widely held assumptions that video and audio automatically add value to Web-based programs [50,51]. Consider, for example, that intriguing research by Reeves and Nass [33] suggests that the value of the verisimilitude of video presentations may be overstated. Their studies suggest that video is not as critical an ingredient in the design of computer-based programs as is the tone of the communication that can be tailored to fit the user's expectations. Fogg [52] echoes this point in his discussion of the persuasive features of the computer as *social actor*. It is reasonable to expect that many users would not be well served by websites in which they become passive page turners while observing multiple “talking head” presentations [53]. Similarly, behavioral researchers have a very significant opportunity to explore and test how best to use the interactivity and tailoring possibilities of Web-based program delivery (eg, [54]).

Future research should consider developing *leanmedia* websites that use vector-based animated graphics to offer interactivity and graphical content without excessive bandwidth consumption. One vector graphics tool to consider in this regard is Macromedia Flash (for its vector graphics capabilities rather than its video capabilities). Flash uses a client-side browser plug-in that has a very large installed base: version 2.0 is on 98.3% of current computers, whereas the most recent version 7.0 is on 90.0% of current computers [55]. Alternatively, it is possible to use an open source structured vector graphics (SVG) tool rather than the proprietary Flash format, but currently only 13% of users have the SVG browser plug-in on their computers [55]. The acceptability of websites may be compromised when users are required to download and install new or updated versions of browser plug-ins, especially in scenarios in which the download is large and users have limited bandwidth, dial-up Internet access.

Researchers and developers of Web-based behavior change programs should consider using intelligent adaptive designs that “sniff out” user’s bandwidth transparently and then tailor program content according to the bandwidth characteristics of each user’s Internet connection. Using this approach would require program designers to consider how best to deliver equivalent content using different media—a topic with both technical and behavior change complexities that have received little empirical study to date.

Website designers should make greater use of Web-enabled CD-ROMs to deliver rich media to the very large base of Internet users who do not have broadband access. Programs of this type would still use centralized functions of a Web-based program (eg, logic, authentication, and data collection), but they would access rich media located on each user’s computer. In this scenario it should be noted that video and audio would not need to be as compressed as for Web delivery. As a result, rich media components could be longer and larger (eg, larger video image), and their enhanced quality could result in better acceptability and improved impact.

Plans to market a Web-based program need to consider the bandwidth demands that might accompany program recruitment announcements or delivery scenarios that encourage concurrent usage. Consider, for example, the possible stress on a smoking cessation website tied to an event that occurs on a single date (eg, the Great American Smokeout) or when an announcement is issued to all employees of a large corporate client. Similarly, consider the scenario in which scores of users are assembled in community computer centers or corporate training rooms to access rich media Web-based programs. Announcements via mass media, email as well as URL publishing, can generate large demand spikes. Resulting peaks of Internet traffic, especially when programs deliver rich media content, can compromise the functioning of any Web-based program.

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Limitations and Strengths of Current Study

There are some noteworthy limitations to the design of the current study. The tests that validated the BUI threshold (number of concurrent users who would have a satisfactory user experience) were not exhaustive. More users could have been included to provide further confidence in the results we report. Nor was it possible for us to qualify the use of BUI by quantifying the peaks and valleys of available bandwidth in different real-world circumstances (eg, phone line degradation [particularly germane of dial-up and DSL users], geographically distributed users, times of day, server and end user hardware/software configurations). It is also important to acknowledge the subjective nature of the “satisfactory user experience” criterion we used. This measure was based upon individual user reactions when accessing program content in a realistic manner. We attempted to operationally define our use cases to be similar to what would typically be experienced by users of these programs.

Our investigation does, however, have a number of strengths. For example, we were able to measure the bandwidth use of a number of Web-based behavior change programs that contained rich media. In addition, we were able to test the impact on bandwidth use of three variations of one of these rich media behavior change programs, which enabled us to recommend the approach of using Web-enabled CD-ROM design to extend the reach of rich media Web programs to the largest possible audience of potential users. Finally, while the BUI is not a precise tool, it nonetheless describes a process that others can replicate in order to evaluate their own Web-based programs. We believe that important implications can be drawn from using the BUI to calculate estimates of the maximum number of concurrent users by Internet access type (Tables 2 and 3).

The Scalability of Behavior Change Websites

Because they attract relatively meager numbers of concurrent users, typical randomized controlled trials of Web-based programs provide an inadequate test of their ability to deliver content in a scalable manner. In some ways, it is as if the airworthiness of a modern airliner is tested with a light load of passengers and fuel in only ideal flying conditions without ever being tested with a full load in windy, stormy conditions. But website scalability is a typical rationale invoked in support of using the Internet as a delivery channel and in justifying the considerable development expenses. We believe it is essential to acknowledge that Web-based programs need to be designed to fit the goals for which they are intended and that their bandwidth requirements need to be considered, measured, and reported in order to gauge how well they will likely operate under their intended use case scenarios.

Conflicts of Interest

None declared.

Multimedia Appendix 1

Calculation of BUI. [PDF file, 80 KB-]

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Abbreviations

BUI: bandwidth usage index

Kbps: kilobits per second

NIH: National Institutes of Health

PHP: PHP hypertext preprocessor

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QoS Requirements of Network Applications on the Internet

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ABSTRACT: In this paper, we present quality of service (QoS) metrics for various network applications based on human factors and technology attributes. The first term, human factors, addresses human perception of different kinds of media, such as conventional text, audio and video. The second term, technology attributes, represents the different technological aspects of these network applications, such as time-dependence and symmetry. Both of these terms are key factors that lead to variations of requirements for QoS. Establishing these requirements is paramount to providing QoS on computer networks and the Internet. With the metrics presented in this paper we can provide the criteria necessary for such QoS assurance.

INTRODUCTION

Information services on the Internet come in varying forms, such as web browsing, e-mail, and multimedia on demand. The main motivation behind the design of next-generation computer and communications networks is providing universal and easy access to these various types of information services on a single multi-service Internet. This means that all forms of communications (video, voice, data and signaling), along with all types of services (from plain text web pages to multimedia applications), are bonded in a single-service platform through Internet technology.

Media are transmitted on a network by bit streams. Each bit stream is represented by a flow of bits of 1s and 0s through the network. Although all media are transmitted in the same digital form, each application generating a bit stream has an associated set of service characteristics. Regardless of the type of media (picture or voice) there are different QoS requirements perceived from the user's standpoint that determine whether the user finds this type of service acceptable. Examples of user perception include phone-to-phone delay and playing music online.

In general a phone-to-phone delay, from a user's perspective, should be no more than 150 milliseconds (Silveira et al, 1996) to allow for appropriate and easy understanding. Outside of this limit, the user will become annoyed and find the service unacceptable. To support this kind of a call over the Internet, the voice media need to be transmitted with a constant and reliable bit transmission rate, low jitter, and a low error and loss rate.

User perception is also important when a user plays music online. If the variation of delay across the Internet is too large the song sounds desultory, and the user stops listening. When a network cannot provide the corresponding support necessary to make a service acceptable, the user feels that this kind of service is senseless.

The Internet today is made up of a simple convergence of joining application media bit streams. This convergence does not take into account the service characteristics associated with each type of

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application. This means that the Internet does not currently provide any quality assurance.

Concise specification of service requirements is vital to realizing quality assurance on computer networks. Since different applications have different service features, each application should specify its explicit requirements to a computer network in order to achieve the desired QoS. If there are no requirements given, the network will take for granted that any level of service is acceptable, and therefore can provide any level of network support. The types of network support provided, such as bandwidth and priority in a router's queue, may lead to delay and jitter that can then render the service unacceptable. Consequently, if we want to satisfy QoS of various applications in next-generation networks, the first step is to identify the QoS metrics of each network application.

In this paper, we first analyze the influence of two key factors, human factors and technology attributes, on QoS requirements. Next, we present network application classification based on the application's different service characteristics, resulting mainly from these two key factors. Finally, based on existing literature, we systematically propose numerical measures, which can quantitatively represent the QoS requirements of various applications with different service characteristics in Multi Service Networks, along with the rationale behind these choices.

QOS MEASURES

In general, QoS has three attributes to measure the output performance of a process: timeliness, precision and accuracy (Ye, 2002). Timeliness measures the time taken to produce the output of the process. Precision measures the amount or quantity of the produced output. Accuracy measures the correctness of the produced output, usually relating to the content of the output. Specific measures of the three QoS attributes depend on the process of interest. Existing work on QoS of computer networks has used the following QoS measures.

Response Time Expected by Users: The users' expected response time is the time elapsed between sending a request and the reception of the first response by the user.

Delay: The network transmit delay is the time elapsed between the emission of the first bit of a data block by the transmitting end-system, and its reception by the receiving end-system.

Jitter: In transmission technology, jitter refers to the variation of delay generated by the transmission equipment.

Data Rate: Data rate refers to the raw data rate of encoded multimedia data before transmission, that is, the rate in which data are encoded.

Required Bandwidth: The required bandwidth is defined by the required data transfer rate, measured in bits per second, of each specific application in telecommunication. This metric includes raw data and overhead.

Loss Rate: The bit loss rate is the number of bits lost between two points in telecommunications after transmission.

Error Rate: The bit error rate is the frequency of erroneous bits between two points in telecommunication after transmission.

TECHNOLOGY ATTRIBUTES OF APPLICATIONS

Different applications can be characterized by certain technology attributes. Applications in one class of technological attribute may not share the same QoS requirements as those in another class. The technology attributes used to classify applications in this paper are defined in this section.

Time Dependence

Applications can be classified by their time dependency requirements. We classify all applications into two time dependant categories: real time (RT) applications and non real time (NRT) applications.

Real time is defined by the Oxford Dictionary of Computing as: “A system in which the time at which the output is produced is significant. This is usually because the input corresponds to some movement in the physical world, and the output has to relate to that same movement. The lag (delay) from the input time to output time must be sufficiently small for acceptable timeliness”.

RT applications can further be divided into soft real time and hard real time. The main difference between soft and hard real time is that hard real time applications will fail should QoS requirements not be met. In this paper we group all soft and hard real time applications as RT.

In RT applications, the network needs to deliver time-based information without changing its built-in time properties. For adequate user satisfaction, we need to maintain more stringent delay and jitter requirements for RT applications. The delay requirements must be strict in order to maintain system timing. The jitter requirements are essential to transmitting data at a constant, reliable rate.

NRT applications are any applications that do not have stringent timing requirements. This type of application does not fail if timeliness metrics are not met, nor does it require timing accuracies to be considered acceptable. NRT applications, which do not have time-based sensitivity requirements, are mostly concerned with delay.

Symmetry

An application's symmetry property allows us to classify all applications into two categories: symmetric applications and asymmetric applications.

In symmetric applications, requests and responses are comparable in terms of resource consumption. Videophony is an example of a symmetric audio and video application. This application requires equal resource consumption on both host machines. In this case, the request consumes an equivalent amount of resources as the response.

In asymmetric applications, requests are considerably less resource consuming than responses. An example of an asymmetric application is Video-on-Demand (VoD). This application is highly asymmetric since it consumes much more resources in the VoD Server (response) than in client machine (request).

The grades of symmetry in multimedia services result in different grades of interactivity, which in turn lead to different response time and delay requirements. For example, Videophony requires an almost instantaneous feedback to satisfy the human perception of conversation, whereas VoD does not. This immediate feedback requires minimal delay and jitter on the network.

HUMAN PERCEPTION OF QOS

In this section we present the human perception of the QoS measures presented previously. These are the perceptions from a user's standpoint, which differ from the perception of these QoS aspects from a technology standpoint.

Human Perception of Delay

In conventional text and data networking, delay requirements are the least stringent. The response time in these types of applications can increase from 2 to 5 seconds before becoming unacceptable (Szuprowicz, 1995). Even at 5 seconds, the response time may still be considered tolerable.

End-to-end audio latency refers to the latency experienced across a link. In other words, the time between when the sound is sent on one end of a link, and when it is received on the other. The relationship between End-to-end Audio Latency and the Human Ear is shown in Table 1. (Szuprowicz, 1995).

In interactive applications of real time sound transmission, as well as in virtual reality, the overall one-way delay needs to be short in order to give the user an impression of real time responses. A maximum value on the order of 0.1 to 0.5 seconds is required to accomplish this goal. (Fluckiger, 1995).

Based on subject tests, the International Telecommunication Unit (ITU) G.114 specification recommends less than a 150 millisecond one-way end-to-end delay for high-quality real time traffic in telecommunication. (ITU G.114, 1996) The G.114 time limits are shown in Table 2.

Table 1
End-to-end Audio Delay and the Human Ear

Audio Delay (ms)	Effect of Delay on Human Voice Perception
> 600	Speech is unintelligible and incoherent
600	Speech is barely coherent
250	Speech is annoying but comprehensible
100	Imperceptible different between audio and real speech
50	Humans cannot distinguish between audio and real speech

Table 2
G.114 Limits for One-way Transmission Time

One-way transmission time	Effect of Delay on Human Voice Perception
0 to 150 ms	Acceptable for most users
150 to 400 ms	Acceptable, but had impact
400 ms and above	Unacceptable

In this paper, we adopt the G.114 limit of delay, 150 milliseconds, for most real time traffic that does not require a high level of interaction, such as teleconferencing, and streaming audio and video. On the contrary, interactive applications like Voiceover IP (VoIP) and Videophony require the value of delay given in Table 1 of 100 milliseconds due to the desire for instantaneous feedback.

In video applications, it is necessary to preserve the timing relationships between audio and video streams, as well as the timing relationships within individual video streams. First, we consider the way in which human beings perceive sounds and images.

Humans are much more sensitive to alteration of audio than of visual signals (Fluckiger, 1995). Lip-synchronization is defined as the difference between the visual reception of a video, and the auditory reception of its sound. For adequate user perception, lip-synchronization should not exceed 100ms (Fluckiger, 1995)

In many applications, an audio and a video stream are transmitted concurrently. Compressor/Decompressor (codec) technology processes audio and video data. In some codec technology, audio and video data are put into separate packets. In such cases, the audio stream must have priority over the video stream. Since the delay of a real time stream should be less than 150 milliseconds, the delay of a video stream should not exceed 250 milliseconds to preserve the 100 milliseconds limit on lip-synchronization. In other codec technologies, audio and video data are put into the same packets. Under these circumstances, we have to satisfy the delay requirement of the audio stream, which has a maximum limit

of 150 milliseconds. In this paper, we consider the case in which audio and video data are in the same packets.

Human Perception of Jitter

Jitter is an essential performance parameter of a network intended to support real time sound and image media. Of all information types, real time sound is the most sensitive to network jitter.

The solution to overcoming jitter in real time sound transmissions is for the receiving system to wait a sufficient length of time, called a delay offset, before playing received sounds. Incoming blocks are stored in a temporary memory location, called a buffer. After an adequate delay, these sound blocks can be played smoothly. This process is called delay equalization or delay compensation.

One of the shortcomings of the delay equalization technique is that of the additional delay, which is introduced at the sink end. Therefore, the value of jitter has a close relationship with the delay requirement of a specific application.

With typical personal computers or workstations as end-systems, the variation of the network transit delay should, in general, not exceed 100ms for CD-quality compressed sound and 400ms for telephone-quality speech. For multimedia applications with strong bounds on transit delay, like virtual reality, this jitter should not exceed 20–30 ms (Fluckiger, 1995).

Video applications have different human perception requirements for the minimization of jitter. For example, the variation of network transit delay should not exceed 50ms for HDTV quality, 100ms for broadcast quality, and 400ms for videoconferencing (Fluckiger, 1995).

Human Perception of Error

Three considerations must be taken into account when defining the acceptable human requirements for error. The first is current retransmission techniques. In conventional text and data transmissions, retransmission techniques are employed to correct errors in transmitted data. In the case of real time audio and video, retransmission techniques are not appropriate. Secondly, we need to realize that humans usually tolerate a high level of transmission error. This tolerance is generally derived from human experiences in comparable situations. Finally, in the case of compressed bit streams, erroneous or lost packets may invalidate greater numbers of subsequent packets. As a result, lower values of network error and loss rates must be maintained when audio and video data are in a higher compression rate scheme.

Furthermore, limits on error rates vary by application. The manner in which data are presented to humans (data type) dictates limits on error rates. In the case where data are presented to human users without recording for further processing, the residual bit error rate of a telephone-quality audio stream should be lower than 10^{-2} . The residual bit error rate of a CD-quality audio stream should be lower than 10^{-3} in the case of an uncompressed format, and lower than 10^{-4} in the case of a compressed format (Fluckiger, 1995).

The end-to-end network bit error rate, before possible error recovery between end-systems, should not exceed 10^{-6} for HDTV quality, 10^{-5} for broadcast TV quality, and 10^{-4} for videoconference quality (Fluckiger, 1995). These figures are for compressed video data streams.

CLASSIFICATION OF APPLICATIONS

Based on the different technology attributes and human perception characteristics discussed in this paper, we propose the taxonomy of traffics on network in this section. The technology attributes

previously defined in this paper serve as the basis for classifying multimedia applications into four categories based on time dependence and symmetry.

Non Real Time & Asymmetric

Non real time, asymmetric applications generate traffic that is considered “best effort” traffic. This means that the traffic is transmitted using a best effort protocol. The best effort protocol does not provide any specific QoS reliability and simply processes traffic on a first-come, first-served basis. The most common NRT, asymmetric network applications are as follows:

- Web Browsing — Hyper Text Transport Protocol (HTTP)
- Enhanced Web Browsing
- Email
- File Transfer Protocol (FTP)
- Telnet.

Non Real Time & Symmetric

Most common NRT network applications are asymmetric. This makes sense because NRT traffic usually consists of client machines requesting data and services from a host machine. However, there is one commonly used NRT application that is symmetric in nature and thus requires equivalent resource consumption at each host endpoint of the communication link:

- Internet Relay Chat.

Real Time & Asymmetric

Real time applications inherently have more stringent QoS requirements due to the nature of real time transmissions. Common real time, asymmetric applications are as follows:

- Audio Broadcasting
- Video Broadcasting
- Interactive Audio on Demand
- Interactive Video on Demand
- Telemetry.

Real Time & Symmetric

Common real time, symmetric applications include those which are conversational in nature:

- Teleconferencing (including Audio, Audiographics, and Video Conferencing)
- Videophony
- VoIP.

QOS METRICS FOR EACH CLASS OF APPLICATION

In this section we systematically propose the desired QoS metrics for each application listed in the previous section. We base these metrics on knowledge about human perception of QoS and the technology features of various traffics put forth in this paper. Capacities of networking equipment are described in Appendices I and II, which can be used to examine the capability of the current networking technology in supporting QoS metrics.

QoS Metrics for Web Browsing

The QoS metrics for conventional web-browsing using HTTP are outlined in Table 3.

Table 3
QoS Metrics for Web Browsing

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Web Browsing	Non Real Time and Asymmetric	2–5 Seconds	< 400	N/A	< 30.5 K	< 30.5 K	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

Requirements for conventional web browsing are mainly influenced by response time, which is limited to no more than 5 seconds. Delay less than 400 ms is expected for all best effort network traffic although, under some circumstances, delays greater than 400 ms may be considered acceptable. Jitter is not applicable in HTTP because it has little impact on traditional static text and picture web browsing. Data rate and required bandwidth for typical applications of web browsing are less than 30.5 kbps. The expected loss rate and error rate are zero since HTTP is a reliable transfer protocol in which erroneous packets are sent again using a retransmission policy.

QoS Metrics for Enhanced Web Browsing

Enhanced web browsing refers to high-priority transaction services, such as those related to e-commerce. The primary performance requirement for enhanced HTTP is to provide a sense of immediacy to the user. This sense of immediacy acts as a security blanket by assuring the user that a transaction is proceeding smoothly. The QoS metrics for enhanced web browsing are outlined in Table 4.

Table 4
QoS Metrics for Enhanced Web Browsing

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Enhanced Web Browsing	Non Real Time and Asymmetric	2–4 Seconds	< 400	N/A	< 24 K	< 24 K	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

For enhanced web browsing, a value of 2–4 seconds for response time is suggested as acceptable to most users. The best effort traffic delay specification of less than 400 ms is sufficient to satisfy this 2–4 second response time requirement. Again, jitter is not applicable to enhanced HTTP. Compared with traditional web browsing, enhanced web browsing requires less bandwidth. This is due to the fact that web pages used in a transaction service are typically simpler and more practical than web pages that are customarily browsed. However, this bandwidth requirement may be increased by adding security measures, such as encryption and SSL. The expected loss rate and error rate are zero.

QoS Metrics for Email

Email is a store and forward type of service which, in principle, can tolerate delays of several minutes or even hours from a technology perspective. However, it is important to differentiate between communications between the user and the local email server and server to server transfers. When a user communicates with a local mail server, there is an expectation that the mail will be transferred quite rapidly, although not necessarily instantaneously. The QoS metrics for email are given in Table 5.

Table 5
QoS Metrics for Email

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Email	Non Real Time and Asymmetric	2–5 Seconds	Low	N/A	< 10 K	< 10 K	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

Consistent with our research findings on response time for web browsing, a requirement of 2–5 seconds for email response time is proposed. The requirement of expected network delay is low, and jitter is not applicable. The typical bandwidth requirement for email is less than 10 kbps. The expected loss rate and error rate are zero.

QoS Metrics for FTP

Typically, FTP service requires a relatively high bandwidth, which is clearly influenced by the size of the file. Meanwhile, the size of files will also impact the time expected to finish the service. We also need to differentiate between the time elapsed between command transmissions, and file transmissions between the user and the FTP server. User expectation is that a file should begin transmission soon after the command sent out. QoS metrics for FTP are given in Table 6.

Again, a requirement for response time of 2–5 seconds is proposed based on our web browsing research findings. As for delay, as long as there is some indication that the file transfer is proceeding, it is reasonable to assume a somewhat greater tolerance for delay than the expected delay for a single web page. Jitter is not applicable, and the expected loss rate and error rate are zero.

Table 6
QoS Metrics for FTP

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
FTP	Non Real Time and Asymmetric	2–5 Seconds	Med	N/A	High	High	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

QoS Metrics for Telnet

Requirements for quality of service of telnet applications are focused on a short delay in order to provide essentially instantaneous echo-back of characters. The QoS metrics for telnet are given in Table 7.

Table 7
QoS Metrics for Telnet

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Telnet	Non Real Time and Asymmetric	< 2 seconds	< 250	N/A	< 1 K	< 1 K	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

For telnet applications, a response time of less than 2 seconds is suggested in order to satisfy the human perception of users and a delay less than 250 millisecond is recommended. Since most of the data transmitted via telnet are plain characters, the requirement for bandwidth is quite small, typically less than 1 kbps. Jitter is not applicable, and the expected loss and error rates are zero.

QoS Metrics for Internet Relay Chat

Internet Relay Chat is a multi-user chat system, where users convene on “channels” (a virtual place, usually with a topic of conversation) to talk in groups, or privately. Requirements for quality of service in Internet Relay Chat are focused on short response times in order to provide essentially instantaneous character feedback. The QoS metrics for Internet Relay Chat are given in Table 8.

Table 8
QoS Metrics for Internet Relay Chat

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Internet Relay Chat	Non Real Time and Symmetric	1 Second	< 200	N/A	< 1 K	< 1 K	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

A response time of less than 1 second is suggested to satisfy users' expectations. A delay less than 200 milliseconds is recommended to guarantee a short response time. In general, Internet Relay Chat consists of plain character transmission, so the requirement for bandwidth is quite small, typically less than 1 kbps. Jitter is not applicable in Internet Relay Chat. With the existence of a retransmission policy for erroneous packets, the expected loss rate and error rate are zero.

QoS Metrics for Audio Broadcasting

User response time in Audio Broadcasting refers to the time elapsed between transmitting commands and receiving feedback from the audio server. The QoS metrics for audio broadcasting are given in Table 9.

Table 9
QoS Metrics for Audio Broadcasting

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Audio Broadcasting	Real Time and Highly Asymmetric	2–5 Seconds	<150	<100	56–64K	60–80 K	<0.1%	<0.1%
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

For audio broadcasting we adopt a one-way end-to-end delay for high-quality real time traffic from ITU G.114 specifications of less than 150 milliseconds. For jitter, no more than 400 ms is recommended for telephone-quality speech and no more than 100 ms is recommended for CD-quality compressed sound (Fluckiger, 1995). We suggest a jitter limit of no more than 100 ms for broadcast quality sound.

A typical data stream for audio broadcasting is 56–64 kbps (Kenyon and Nightingale, 1992). The corresponding bandwidth requirement is around 60–80 kbps. The residual bit error rate is recommended to be lower than 10^{-2} for a telephone-quality audio stream and lower than 10^{-4} for a CD-quality audio stream using a compressed format (Fluckiger, 1995). Since there is a higher quality requirement in

broadcasting than in telephone, and a lower quality requirement than in a CD audio stream, we recommend a loss rate and error rate of no more than 10^{-3} for broadcast audio streams.

QoS Metrics for Video Broadcasting

In Video Broadcasting, under different codec technologies and quality requirements, there are different requirements for network transmissions. We list the QoS metrics for video broadcasting using some typical codec technologies in Table 10.

Table 10
QoS Metrics for Video Broadcasting

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	n/a	Required Bandwidth (bps)	Loss Rate	Error Rate
Video Broadcasting	Real Time and Highly Asymmetric	2–5 Seconds Lip-synch: <100ms	<150	<100				
Typical Application		Coding Standard						
VCR Quality		MPEG-1		<100		1.2–1.5M	<0.001%	<0.001%
Video Quality slightly superior to that of broadcast TV (NTSC or PAL) with bit rate of 4M		MPEG-2				4–60 M		
HDTV requiring bit rate from 15–34 M				<50			<0.0001%	<0.0001%
Multimedia on Web		MPEG-4		<150		28.8–500K	<0.001%	<0.001%
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

For VCR quality MPEG-1 video streams, the required bandwidth is around 1.2–1.5 Mbps (Silveira et al, 1996). Jitter is recommended to be less than 100 ms for a broadcast-quality video stream (Fluckiger, 1995). The residual bit error rate is recommended to be lower than 10^{-5} for a broadcast TV quality video stream using a compressed format (Fluckiger, 1995). The loss and error rates should be less than 10^{-5} .

For HDTV quality MPEG-2 video streams, the required bandwidth is around 40 Mbps (Silveira et al, 1996). Jitter is recommended to be less than 50 ms for HDTV quality video streams (Fluckiger, 1995). The residual bit error rate is recommended to be lower than 10^{-6} for HDTV quality video streams using a compressed format (Fluckiger, 1995). The loss rate and error rate should be less than 10^{-6} .

For typical MPEG-4 video streams on the World Wide Web, the required bandwidth is around 28.8–500 kbps (Silveira et al, 1996). Jitter less than 150 ms is recommended due to a lower quality requirement than VCR technologies, which require a jitter less than 100ms. On the other hand, MPEG-4 also has a higher compression rate, which usually requires less residual error. Based on considerations of quality and compression rate, we recommend a loss rate and error rate less than 10^{-5} .

QoS Metrics for Interactive Audio on Demand

Interactive audio-on-demand on the World Wide Web is the ability to listen to sound on a personal computer (with a sound card and speakers installed) while the file is passed through an individual modem. Under different codec technologies and quality requirements, there are different requirements for network transmission. We list the QoS metrics for some typical audio-on-demand applications in Table 11.

Table 11
QoS Metrics for Interactive Audio on Demand

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	n/a	Required Bandwidth (bps)	Loss Rate	Error Rate
Interactive Audio on Demand	Real Time and Highly Asymmetric	2–5 Seconds	<150	<100				
Typical Application		Coding Standard						
MP3, 64K for mono, 128K for stereo		MPEG-1			32–448K	<0.1%	<0.1%	
Advanced Audio Coding (AAC) providing CD quality audio		MPEG-2			384K	<0.01%	<0.01%	
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

For a mono-quality MP3 audio stream, the required bandwidth is 64 kbps. For a stereo-quality MP3 audio stream, the required bandwidth is 128 kbps. Delay less than 150ms is recommended for high-quality real time traffic in the ITU G.114 standard (ITU, 1996). A jitter rate of less than 100 ms for CD-quality compressed sound is recommended (Fluckiger, 1995). The residual bit error rate is recommended to be lower than 10^{-3} for a broadcasting-quality audio stream in a compressed format. Therefore, the loss rate and error rate should be less than 10^{-3} .

For a CD-quality MP2 audio stream, the required bandwidth is 384 kbps. A delay less than 150ms is recommended for high-quality real time traffic in the ITU G.114 standard (ITU, 1992). A jitter rate of less than 100 ms for CD-quality compressed sound is recommended (Fluckiger, 1995). The residual bit error rate is recommended to be lower than 10^{-4} for a CD-quality audio stream in a compressed format. Therefore, the loss rate and error rate should be less than 10^{-4} .

QoS Metrics for Interactive Video on Demand

Interactive video-on-demand refers to a range of applications whereby users can request access to video servers, which provide still and moving pictures on an individual basis (Fluckiger, 1995). Elapsed time in video-on-demand refers to the time elapsed from a request generation to the start of video playback. The QoS metrics for video-on-demand are given in Table 12.

Elapsed time of a few minutes should be acceptable for video-on-demand because users are not particularly concerned with the starting delay, but instead are concerned with the delay during playback. Response to interactive functions, such as play and next, should be less than 2–5 seconds. The typical

bandwidth requirement for the user's local network is 1.5 Mbps. The bandwidth requirement for the backbone depends on how many users the video server needs to support simultaneously. In general, backbone bandwidth is hundreds of megabits per second.

Other QoS metrics for video-on-demand are the same as those for video broadcasting.

Table 12
QoS Metrics for Interactive Video on Demand

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	n/a	Required Bandwidth (bps)	Loss Rate	Error Rate
Interactive Video on Demand	Real Time and Highly Asymmetric	Elapsed Time: a few minutes Response to Interactive function: 2–5 Seconds Lip-synch: <100ms	<150			Local: 1.5M (typical application) Backbone: hundreds of mega		
Typical Application		Coding Standard						
VCR Quality		MPEG-1		<100		1.2–1.5M / A Single Video	<0.001%	<0.001%
Video Quality slightly superior to that of broadcast TV (NTSC or PAL) with bit rate of 4M		MPEG-2				4–60 M / A Single Video		
HDTV requiring bit rate from 15–34 M				<50			<0.0001 %	<0.0001 %
Multimedia on Web		MPEG-4		<150		28.8–500K / A Single Video	<0.001%	<0.001%
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

QoS Metrics for Telemetry

Telemetry refers to the telemetering and telecontrol of industrial processes. Telemetry is an example of a data service which requires real time streaming performance. Clearly, two-way control implies a tight limit on the allowable delay in telemetry. The QoS metrics for telemetry are given in Table 13.

For telemetry, a delay value of 250 milliseconds is proposed (QoS Performance Requirement for UMTS (Nortel, 2001)). The range of bandwidth requirements is between 2 kbps and 52 Mbps, depending on the specific telemetry system. A key differentiator from voice and video services in this category is the zero tolerance for information error and loss (QoS Performance Requirement for UMTS (Nortel, 2001)). It is clearly understandable that there cannot be data error and loss when controlling an important industrial process.

Table 13
QoS Metrics for Telemetry

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	n/a	Required Bandwidth (bps)	Loss Rate	Error Rate
Telemetry	Real Time and Highly Asymmetric		<250	<100		2K–52M	Zero	Zero
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

QoS Metrics for Teleconferencing

Teleconferencing can occur in three forms: audio conferencing, audiographics conferencing, and video conferencing. Each of these forms requires its own QoS metrics. The metrics for the three forms of teleconferencing are outlined in this section.

QoS Metrics for Audio Conferencing

Audio Conferencing provides an audio link similar to that of a conventional telephone, except that it offers much higher quality audio and enables more than two sites to be linked together. Using hands-free audio units with sensitive microphones and sophisticated echo-cancellation software, audio-conferencing enables communication between groups of participants rather than just individuals (Bleazard, 1985). Voice data transmission dictates the QoS metrics of audio conferencing shown in Table 14.

Table 14
QoS Metrics for Audio Conferencing

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Audio Conferencing	Real Time and Symmetric		<150	<400			<1%	<1%
Coding Standard								
G.711					64 K	80 K		
G.726					40~16 K	50–22 K		
G.728					16 K	22 K		
G.729					8 K	11 K		
G.723.1					6.3/5.3 K	9 / 8 K		
GSM FR					13 K	18 K		
GSM EFR					12.2 K	17 K		
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

ITU G.114 specification recommends less than a 150 millisecond one-way end-to-end delay for high-quality real time traffic, such as voice transmissions. Jitter of no more than 400 milliseconds is recommended for telephone-quality speech (Fluckiger, 1995). In the case of real time audio and video, retransmission techniques are not appropriate. G.729 codec requires packet loss far less than 1 percent to avoid audible errors. In the case of presentation to human users without recording for further processing, the residual bit rate error rate of a telephone-quality audio stream should be lower than 10^{-2} (Fluckiger, 1995). Differing Codec technologies lead to different bandwidth requirements.

QoS Metrics for Audiographics Conferencing

Except for the audio link, audiographics conferencing enables participants at two or more sites to have a shared workspace on their computer desktops. This might be a shared “whiteboard” where they can draw, write, or import and manipulate images collaboratively in real time. Audiographics conferencing might also be “application sharing”, where a piece of software can be run and controlled by both users (G B Bleazard, 1985). This type of conferencing is useful when users at different sites want to work together on documents such as reports or statistical data in spreadsheets. The largest QoS concern for audiographics is voice quality. The QoS metrics for audiographics conferencing are given in Table 15.

Table 15
QoS Metrics for Audiographics Conferencing

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		n/a	Delay (ms)	Jitter (ms)	n/a	Required Bandwidth (bps)	Loss Rate	Error Rate
Audiographics Conferencing	Real Time and Symmetric		<150	<400		9.6–19.6K	<1%	<1%
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

ITU G.114 specification recommends less than a 150 millisecond one-way end-to-end delay for high-quality real time traffic, such as voice. For jitter, a rate of no more than 400 milliseconds is recommended for telephone-quality speech (Fluckiger, 1995). The G.729 codec requires a packet loss rate of far less than 1 percent to avoid audible errors. In the case of presentation to human users without recording for further processing, the residual bit error rate of a telephone-quality audio stream should be lower than 10^{-2} (Fluckiger, 1995). Typical bandwidth requirements for audiographics conferencing are 9.6–19.6 kbps (Bleazard, 1985).

QoS Metrics for Video Conferencing

Video conferencing enables real time communication over a distance by allowing people at two or more sites to communicate with each other. In video conferencing, in addition to hearing each others voices, as with a conventional telephone, people can see video pictures of each other from their various sites (Janssen et al, 2001). Each site has one or more cameras, microphones, loudspeakers and monitors, as well as a codec. This type of conferencing aims to create a sense of people from distant sites being in the same room, an effect that has been called Virtual Presence. Video conferencing is not only concerned with voice quality, but is also concerned with video quality. The QoS metrics for video conferencing are given in Table 16.

Table 16
QoS Metrics for Video Conferencing

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Video Conferencing	Real Time and Symmetric	Lip-synch: <100 ms	<150	<400			<0.01%	< 0.01%
Coding Standard								
H.320					64–1920K	80K–2M		
H.323					64X K	80X K		
H.324					<64K	<80K		
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

Since humans are much more sensitive to alteration of audio signals than visual signals, we adopt the recommendations for delay and jitter from ITU G.114: less than a 150 millisecond one-way end-to-end delay for high-quality real time traffic and no more than a 400 millisecond jitter for telephone-quality speech (Fluckiger, 1995). However, the requirement for loss rate and error rate differ. Recall that compressed streams are more sensitive to errors than uncompressed streams since one lost packet may influence a number of subsequent packets. Similarly, streams with a higher compression ratio are more sensitive to errors than streams with a lower compression ratio since more packets will be impacted. Therefore, the end-to-end network bit error rate for compressed streams, before possible error recovery, between end-systems is recommended not to exceed 10^{-4} for videoconference quality (Fluckiger, 1995).

The H.320-series governs the basic video-telephony concepts of audio, video and graphical communications by specifying requirements for processing audio and video information, providing common formats for compatible audio/video inputs and outputs, and defining protocols that allow a multimedia terminal to utilize the communication links and synchronization of audio and video signals. H.320 standards address ISDN Videoconferencing. The H.323 standards address Video communications on Local Area Networks, and the H.324 standards address video and audio communications over low bit rate connections such as POTS modem connections. From Table 16, we can see that different technology standards lead to different bandwidth requirements.

It is worthy to note that the bandwidth requirements for video conferencing and videophony (described in the next section) may be less stringent than that of general video. This is based on the assumption that these media will likely have less motion and little or no scene changes in the picture.

QoS Metrics for Videophony

Videophony is a companion audiovisual service of videoconferencing. It may be distinguished by the following features (Kenyon and Nightingale, 1992): it is primarily for person-to-person rather than group-to-group audiovisual communication, and it is usually an on-demand service provided on customer-switched networks, whereas videoconferences can be scheduled in advance. Compared with video conferencing, videophony is a more interactive application, which requires instantaneous responses from dialogists. The QoS metrics for videophony are given in Table 17.

Videophony requires a shorter end-to-end delay of 100 ms to make the difference between audio and real speech imperceptible to users (Fluckiger, 1995). Beyond 100 ms, although speech is still comprehensible, users may become irritated with the service. Since videophony is a special application of video conferencing, it complies with the same technology standards and has the same QoS metrics as video conferencing.

Table 17
QoS Metrics for Videophony

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Videophony	Real Time and Symmetric	Phone to Phone Delay: <150 ms Lip-synch: <100 ms	<100	<400			<0.01%	< 0.01%
Coding Standard								
H.320					64–1920K	80K–2M		
H.323					64X K	80X K		
H.324					<64K	<80K		
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

Table 18
QoS Metrics for Voiceover IP

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Voiceover IP	Real Time and Symmetric	Phone to Phone Delay: <150 ms	<100	<400			< 1%	< 1%
Coding Standard								
G.711					64 K	80 K		
G.726					40~16 K	50–22 K		
G.728					16 K	22 K		
G.729					8 K	11 K		
G.723.1					6.3/5.3 K	9 / 8 K		
GSM FR					13 K	18 K		
GSM EFR					12.2 K	17 K		
Network Capacity								
Link: Refer to Appendix I				Router: Refer to Appendix II				

QoS Metrics for Voiceover IP

The QoS metrics for Voiceover IP are given in Table 18.

Like videophony, VoIP also requires a shorter end-to-end delay of 100 ms to give the users an imperceptible difference between audio and real speech (Szuprowicz, 1995). A jitter rate of less than 400 milliseconds is suggested for telephone-quality speech (Fluckiger, 1995). We adopt the loss rate requirement, packet loss far less than 1 percent, in the G.729 codec. In the case of presentation to human users without recording for further processing, the residual bit rate error rate of a telephone-quality audio stream should be lower than 10^{-2} (Fluckiger, 1995). Again, different voice coding technologies lead to different bandwidth requirements.

SUMMARY

In order for users to find a certain type of multimedia service acceptable, the application must meet specific QoS requirements. Metrics used to determine QoS include response time, delay, jitter, data rate, required bandwidth, loss rate and error rate. In order to provide QoS on a network, requirements for QoS must be established. Defining the limits of QoS metrics helps to establish necessary requirements.

In this paper, we consider two key factors of QoS requirements: human factor and technology attributes. Based on our survey of human factors and technology features, we propose numeric QoS metrics of network applications in this paper. Up to date, we have not found any literature stating metrics which differed from our findings. Based on the lack of literature which would dispute these findings, we could not definitively state whether or not specific measures are “widely accepted”. However, we believe that metrics proposed in this paper provide a starting point to establish a universally accepted set of QoS requirements for network applications on the Internet. We concede that these proposed measures could be refined through experimental verification. Further work can make use of these metrics to differentiate network traffic for realizing quality assurance in networks.

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Appendix I: Link Capacity of Networks

Wide Area Networks (WANs)

Technology used to build WANs	Speed
X.25	45 M bps–several G bps
ISDN	
Frame Relay	
SMDS (Switched Multi-megabit Data Service)	
ATM	

Local Network (LANs)

Technology used to build LANs	Speed	Topology	Medium	Access Mechanism
Ethernet	10 M bps	Bus	Twisted Pair	CSMA/CD (Carrier Sense with Multiple Access / Collision Detection)
	100 M bps			
Token Ring	4 M bps	Ring	Shielded Twisted Cable	Token-passing
FDDI	16 M bps			
(Fiber Distributed Data Interconnect)	100 M bps	Ring (Two rings)	Optical Fiber	Token-passing

LAN to WAN

Connection	Speed
Dedicated leased line at T1	1.544 M bps
Dedicated leased line at T3	45 M bps

Home to WAN

Connection	Speed
Dial-up Analog Modem	14.4 ~ 56 K bps
ISDN Basic Rate Interface (BRI)	128 K bps
ISDN Primary Rate Interface (PRI)	1.544 M bps
High-Bit-Rate Digital Subscriber Line (HDSL)	1.544 M bps
Asymmetric Digital Subscriber Line (ADSL)	Outgoing: 640 K bps; incoming: 6 M bps
Cable TV Companies	10 M bps

Appendix II: — Router Capacity of Network

Router	Upper Bandwidth
Cisco 12000 Series Internet Router	10 G bps
Cisco 10700 Series Internet Routers	5 G bps
Cisco 10000 Series Internet router	51.2 G bps
Cisco 7600 Series Internet Router	256 G bps
Cisco 7500 Series Internet Router	2 1.067 G bps
Cisco 7300 Series Internet Router	16 G bps
Cisco 7200 Series Internet Router	1.2 G bps
Cisco 7100 Series Internet Router	140 M bps