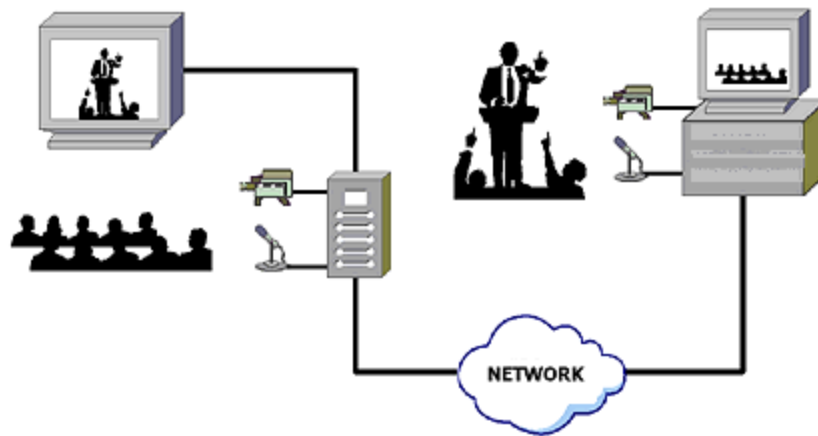


Training Manual

On



IP based Videoconferencing

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Chapter I

Videoconferencing Basics

Topics Covered

- Introduction to Videoconferencing
- Point-to-Point Videoconferencing
- The Gatekeeper
- Multipoint Videoconferences
- Streaming
- Gateways
- Bandwidth Considerations

Introduction

With the ever-increasing reach of the Internet, corporations are taking advantage of its flexibility and cost-effectiveness to expand and enhance their traditional IP-based enterprise networks and Applications. This increases demand for greater capacities across the Internet.

It has also been discovered that the value of IP-based, real-time voice and video applications is massive.

As these applications become more commonplace, network managers are creating networks that overlay voice and video applications on top of the traditional enterprise network.

When real-time, media-rich applications are merged with traditional enterprise applications and networks, several key issues must be addressed, including latency, jitter, security, NAT and firewalls.

Videoconferencing Definition and Overview

What is videoconferencing?

Videoconferencing is defined as:

"Conducting a conference between two or more participants at different sites by using computer networks to transmit audio and video data. For example, a point-to-point (two-person) video conferencing system works much like a video telephone. Each participant has a video camera, microphone, and speakers mounted on his or her computer. As the two participants speak to one another, their voices are carried over the network and delivered to the other's speakers, and whatever images appear in front of the video camera appear in a window on the other participant's monitor."

"Multipoint videoconferencing allows three or more participants to sit in a virtual conference room and communicate as if they were sitting right next to each other. Until the mid 90s, the hardware costs made videoconferencing prohibitively expensive for most organizations, but that situation is changing rapidly. Many analysts believe that videoconferencing will be one of the fastest-growing segments of the computer industry in the latter half of the decade."

Expanding this definition slightly to include not only computer network-based videoconferences but those that utilize ISDN telecommunications links as well.

Also, the multimedia nature of modern videoconferencing technology provides for more than just audio/video conferencing. It provides for true multimedia conferencing that includes the sharing of data between applications as well. Lastly, we would like to emphasize the interactive nature of videoconferencing.

Thus we might say: Videoconferencing is

"Conducting an interactive conference between two or more participants at different sites by using computer networks or ISDN telecommunication links to transmit audio, video, and data. "

Videoconferencing Protocols

Standardization

In order to ensure that systems are able to work with each other independent of brand and or system type, a number of standards have been developed by the International Telecommunications Unit (ITU). You'll see and hear these standards referred to by different equipment manufacturers when describing their products.

The following are ITU Videoconferencing standards.

The Umbrella Protocol H.323 provides specifications for voice and video communication over an IP network.

H.320 provides specifications for voice and video communication over ISDN lines.

H.324 provides specifications for voice and video communication over traditional phone lines.

Comparison of Protocols:

H.323 versus H.320

As described above, there are a number of videoconferencing technologies in use. In this presentation we shall concentrate on two that are in popular: H.320 and H.323-based videoconferencing technologies.

H.320

This is a protocol that defines how real-time multimedia communications and conferencing are handled over switched or dedicated ISDN telecommunication links.

The protocol is an international standard of the International Telecommunications Union (ITU), and it was adopted in 1990. Multimedia refers to the fact that the standard covers voice, video, and data. The standard is an umbrella standard and includes many other protocols that describe, as an example, how to encode and decode voice and data, how to setup calls between terminals, and how to handle data connections.

The above definition of H.320 is rather complex. Lets see if we can describe it in simpler terms. Lets assume we have a H.320 compliant terminal that has a microphone, a speaker system, a display, a camera, an ISDN connection to the public telephone network, and the necessary electronics to implement the H.320 protocols. Assume another similar terminal exists at a remote site.

The local user can dial the ISDN telephone number of the remote terminal, and the H.320 protocols handle the call setup between the terminals. The local terminals microphone and camera picks up the audio and video from the local user, decodes and compresses the audio/video stream, and sends it to the remote site all using the protocols as defined in the H.320 standard. The digital stream is transmitted via the ISDN telecommunication lines to the remote site where it is uncompressed, decoded, and displayed and heard by the remote user.

A similar audio/video stream is formed at the remote site and sent to the local site where the audio and video from the remote site can be heard and seen by the local user. A fully interactive videoconference can thus be held between the two sites.

H.320-based videoconferencing is considered to be "traditional", and has been around for a number of years. The problem is that H.320 terminals are expensive, and they normally are implemented in videoconferencing rooms that are, in themselves, very expensive to construct. Vendors have not made H.320 equipment that is applicable to desktop videoconferencing. In addition, ISDN telecommunications lines are expensive to install and incur rather large operating costs. However, there are a lot of H.320 rooms still in use, and thus H.320 is still an important part of the videoconferencing implementations over the world.

H.323

With the advent of the Internet, a new low-cost communication medium became available for communication between sites both local and remote users. Perhaps we should qualify this and say that at least the incremental cost to the end user to use the Internet is low-cost. There are generally no usage costs for the Internet, the bandwidth, at least between potential user sites, is quite high, and the reliability is very good. This motivated developers to create a videoconferencing technology that uses the Internet to interconnect sites. H.323 is the resulting protocol they developed.

H.323 is an umbrella set of standards defining real-time multimedia communications and conferencing over packet switched networks.

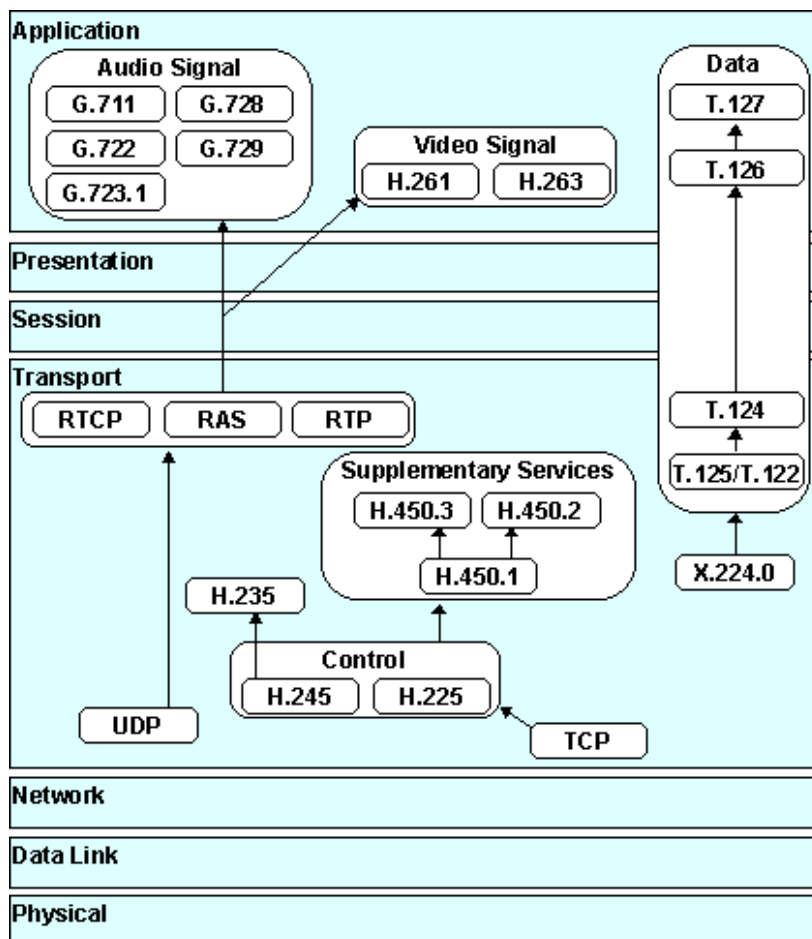


Fig1. H.323 standard as defined in the OSI layer.

H.323 Version 1 was adopted by the ITU in 1996 and Version 2 was finalized in 1998. Like H.320, it is a part of a family of H.32X standards. H.323 borrowed many of the H.32X standards used by H.320, especially those for encoding/decoding the audio/video streams and the data sharing protocol (T.120), but it defined a new set for handling communication over the Internet.

Like H.320, vendors have produced a wide array of compatible H.323 compliant products. Unlike H.320, the H.323 products cover a much wider spectrum of capabilities.

Videoconferencing has thus moved into the "do it yourself" category and no longer requires expensive videoconferencing rooms. This low entry cost for equipment, the ease of interconnecting H.323 equipment over the Internet, and the fact that equipment exists that allows H.320 and H.323 terminals to interoperate in the same videoconference has led to a rapid deployment of H.323 equipment for videoconferencing in the educational environment.

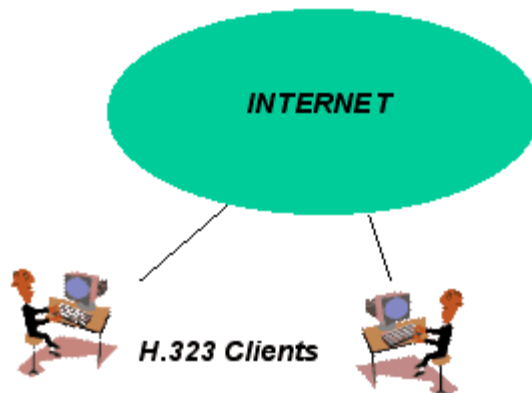
Types of Videoconferencing

H.323 is an International standard protocol for videoconferencing. It uses the Internet for connectivity between endpoints. Endpoints can be client videoconferencing terminals, Multipoint Control Units (MCUs), or gateways. This following section describes the various endpoints and how they interoperate.

Point-to-Point Videoconferencing

Consider two client terminals that are connected to the Internet. ([See Figure 2](#)) An example of a client terminal or end point is a VCON end point Falcon IP or Polycom's Viewstation SP. This terminals and its associated peripherals allow the user to make a call to another client, send the local audio/video stream to the remote client, and hear/view the received audio/video stream on a local speaker/monitor that is connected to it.

Figure2. Point-to-point VC

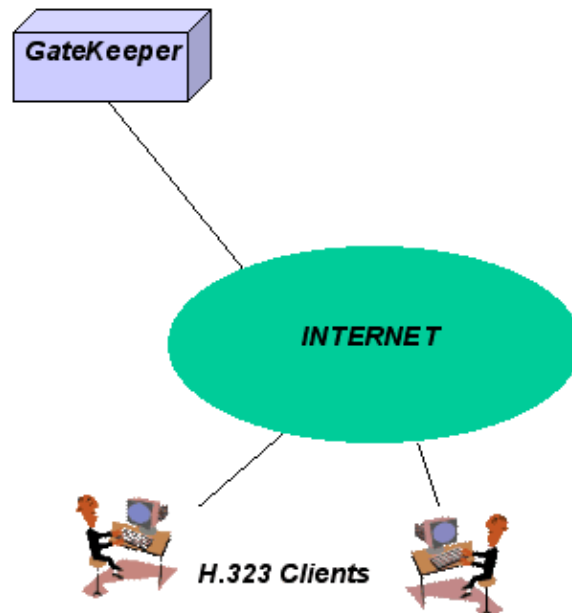


Assume one user (the local user) uses a Falcon IP to call a user at a remote Falcon IP (client terminal) by entering the IP address of the remote Falcon IP. The clients setup a call between the stations following the specifications of the H.323 protocol. Once the call is setup, the clients exchange audio/video streams over the Internet. The point-to-point videoconference continues until one of the users "hangs up" the call.

One of the problems with this type of video call is that IP numbers are used for the call. IP numbers are difficult to remember; some users have dynamically assigned (DHCP) IP numbers that can change every time they boot their system; there are problems in using IP addressing when different vendor systems are used. Although it is occasionally used, the use of IP dialing for creating videoconferencing sessions is not recommended.

To alleviate the problem of IP dialing, the H.323 standard defines the use of a gatekeeper. [\(See Figure 3\)](#) The gatekeeper is a system that connects to the Internet just like the client terminals. The IP address of the gatekeeper is configured into the client terminals and when the clients "power up", they communicate with the gatekeeper and transfer certain information to the gatekeeper that describes the client. This process is known as registration; the client registers with the gatekeeper.

Figure 3. Use of gatekeeper



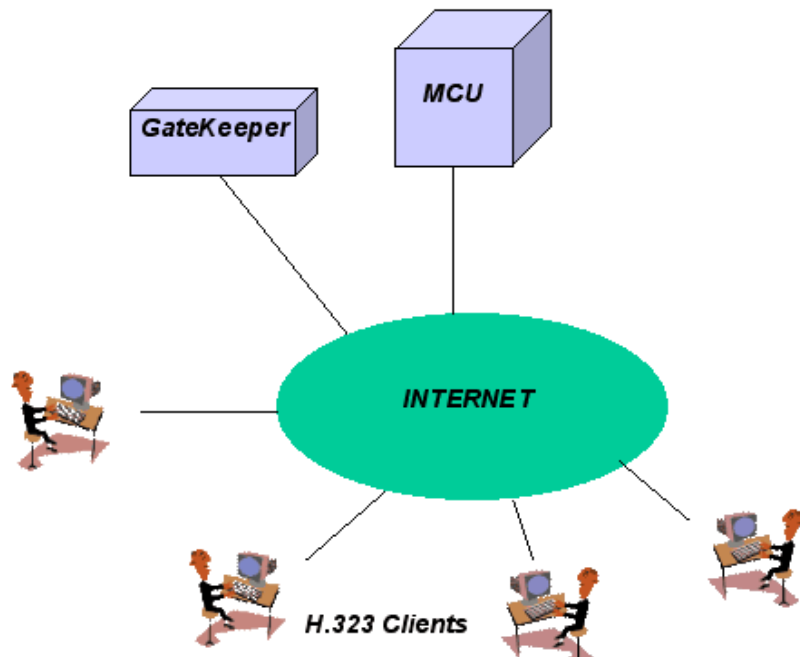
Two identifiers are assigned to and configured in each client terminal. One is a H.323 Alias. It is usually descriptive of the particular client terminal and usually contains alphanumeric characters. The other descriptor is the H.323 Extension. It usually consists of several numbers and can be thought of as being the video telephone number of the client. While it is possible to use either the H.323 Alias or the H.323 extension for dialing, it is difficult to dial alphanumeric characters on most clients; it is the H.323 extension that is normally used for dialing.

When the clients register with the gatekeeper, they pass their IP numbers, H.323 alias, and H.323 extension to the gatekeeper where it is stored. This allows a local user to dial a remote user by entering the remote users H.323 extension (video telephone number) rather than an IP address. The local client terminal communicates the H.323 extension to the gatekeeper. The gatekeeper then checks to see if the remote client is registered with the gatekeeper. If it has, the gatekeeper sets up the call between the two clients; if it is not registered, the call is rejected. Once the call has been setup, the audio/video streams flow directly between the clients over the Internet. The gatekeeper can perform a number of other management functions as well.

Multipoint Videoconferences

To this point we have only considered point-to-point videoconferences. These are conferences between two client terminals. The question can then be raised, "what if we have users at three or more clients that want to hold a videoconference". To handle this situation, the H.323 standard introduces the concept of a Multipoint Control Unit (MCU). The MCU ([See Figure 4](#)) is an endpoint that can be thought of as a "video bridge". The MCU connects to the Internet as does any other endpoint and registers with the gatekeeper, as does any other endpoint.

Fig 4. Multipoint Videoconferences



MCU, depending on its design capacity, can handle a certain number of

simultaneous videoconferences each with each videoconference being logically separate from the others and with each having a specified number of users. System administrators define "services" on the MCU where each service has certain characteristics that contrast it from other defined services on the MCU. As an example, a service of 75 to 700 might be defined that allows for several simultaneous videoconferences to be created where each have a maximum size of, say, five sites (clients) and where all must encode their audio/video streams at 384 Kbps. A specific videoconference on service 75 is then defined by the service number and by a conference "password" (e.g. 751234). Each of the simultaneous videoconferences that are held on service 75 is then defined by the service number (75) and by a different password.

When users want to join a particular videoconferencing session, they dial the service number/password combination. The gatekeeper checks to see if that service has been registered by a MCU. If it has, the gatekeeper completes the call by connecting the client to the specified videoconference on the MCU; if the service has not been registered, the call is rejected.

Once the call has been connected, the client's audio/video stream is then sent over the Internet from the client to the MCU. Similarly, other clients connect to the session and send their audio/video streams to the MCU. The MCU selects one of the audio/video streams on the videoconference and returns that audio/video stream to all of the clients (that is all except the client whose stream was selected).

There are several methods for selecting an audio/video stream. Audio switching and chairman control are two alternatives. Typically, the method that is chosen is audio switching where the MCU selects the stream that currently has active audio (someone is talking or is talking the loudest). We frequently refer to this selection process by saying that this particular stream (client) has "captured" the MCU.

Lets assume that we have several clients connected to a single videoconferencing session on a MCU. The assumption is that no users want to have the MCU send them back video of themselves and no site wants to receive an audio stream that contains their own audio. So the MCU sends the selected video stream to all the clients except the client whose stream was selected; the MCU sends the video from the last site that was selected to the currently selected site. All of the audio streams are aggregated together and sent back to each site except

with their audio removed. Thus each site gets a unique audio stream. Each stream only contains the audio from the other sites.

As the user(s) at one site stop talking and the user(s) at another site start to talk, they capture the MCU. The process is repeated with the video from the newly selected site now being sent to all the other sites, and the newly selected site getting the video from the previously selected site.

Streaming

To participate in a H.323 videoconference, users must have appropriate videoconferencing client terminals and have Internet connectivity with sufficient bandwidth to support the videoconference.

Some users may not have these capabilities but would still like to be able to participate even if that meant that they could only see and hear conference participants but not be able to interact with them. This can be accomplished if the videoconference session is captured, encoded in an appropriate format, and streamed over the Internet although this capability is not a part of the H.323 standard.

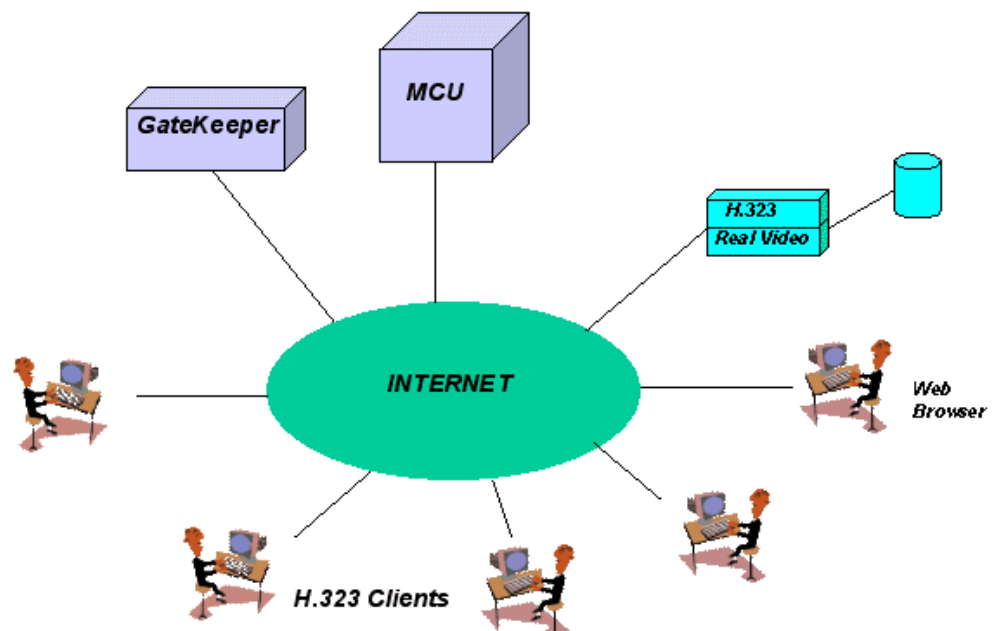


Figure 5
streaming

To accomplish the streaming, a H.323 client must be connected to the conference session to be streamed. This station will be able to capture and decode the audio/video that the MCU has currently selected. This decoded audio/video can then be re-encoded and streamed over the Internet.

There are two popular encoding standards that are currently being used: Real Video and Microsoft Windows Media.

The encoded audio/video can then be either streamed on the Internet by a server or archived on a disk file for later viewing or both. The system consists of a H.323 client, an encoder, a server, and an archive storage system.

Users can receive the stream using a browser on a computer. They enter the URL of the server, and the server starts the encoded audio/video stream over the Internet to the computer. Plug-Ins for the browser exist that are capable of decoding both Real Video and Windows media streams. The user can thus see and hear the participants in the streamed videoconference in near real-time. Alternatively, a user can connect to the server at a latter date and view the archived version of the videoconference.

IP Multicasting

IP Multicasting is a new technological phenomenon that contributes a great deal of investment benefits in delivering videoconferencing sessions to multiple users. It is a way of delivering content to multiple users by utilizing the bandwidth only once at a time.

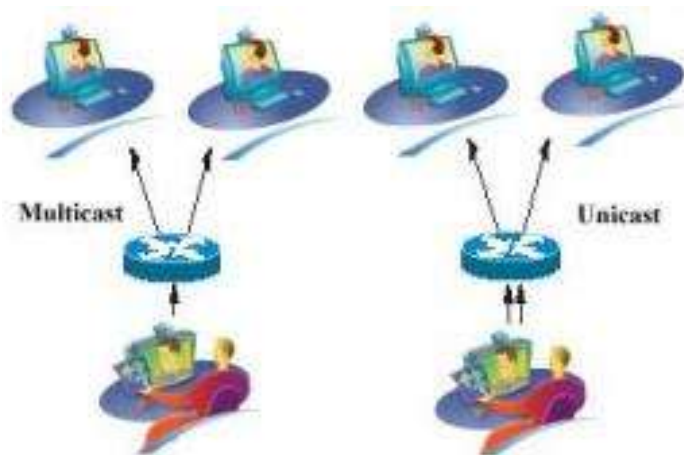


Fig 6. IP Multicasting.

It provides more bandwidth economy than unicast content delivery.

Based on TCP/IP multicast theory, using a common class D multicast group address, an initiating multicast system (Chairman) is able to call multiple endpoints and broadcast audio and video data to them simultaneously. Unlike unicast transmission, which would require central MCU hardware or software, multicast operates with single streams of video and audio, reducing the overall bandwidth required for the conference.

In a unicast call, between two endpoints on the network, the amount of bandwidth utilized must be doubled, as there is video and audio data being transmitted by both endpoints simultaneously. If this conference is expanded (through centralized MCU hardware or software), the bandwidth used on the network increases by a factor of 2 for each endpoint added to the conference.

With multicast, the video and audio data is a one-way transmission, and therefore, the overall bandwidth of the conference never changes, regardless of the number of endpoints that join.

Interactive Multicast takes this theory one step further - although multicast is very useful for one-way communication, by adding small, direct unicast channels between the receiving stations and the chairman, it is possible to send requests and messages between participants. This communication 'back channel' thus provides us with a control method for switching off the multicast at one station and switching it on at another, effectively switching the source of transmission amongst participants as required.

For instance VCON's Interactive Multicasting technology is an exclusive software package available for most VCON LAN end-points and provides the ability to broadcast audio and video streams from any Interactive Multicast enabled end-point on the network.

Videoconferencing among different protocols

So far we have discussed H.323 videoconferencing capabilities. However, many sites have videoconferencing rooms that implement the H.320 standard that uses telecommunication lines (e.g. dial-up or dedicated ISDN lines).

Gateways

H.323 standard was developed after the H.320 standard and uses many of the encoding/decoding protocols originally developed for H.320. The H.320 systems can be considered to be legacy systems, but since many of them still exist, it is important that we continue to support H.320.

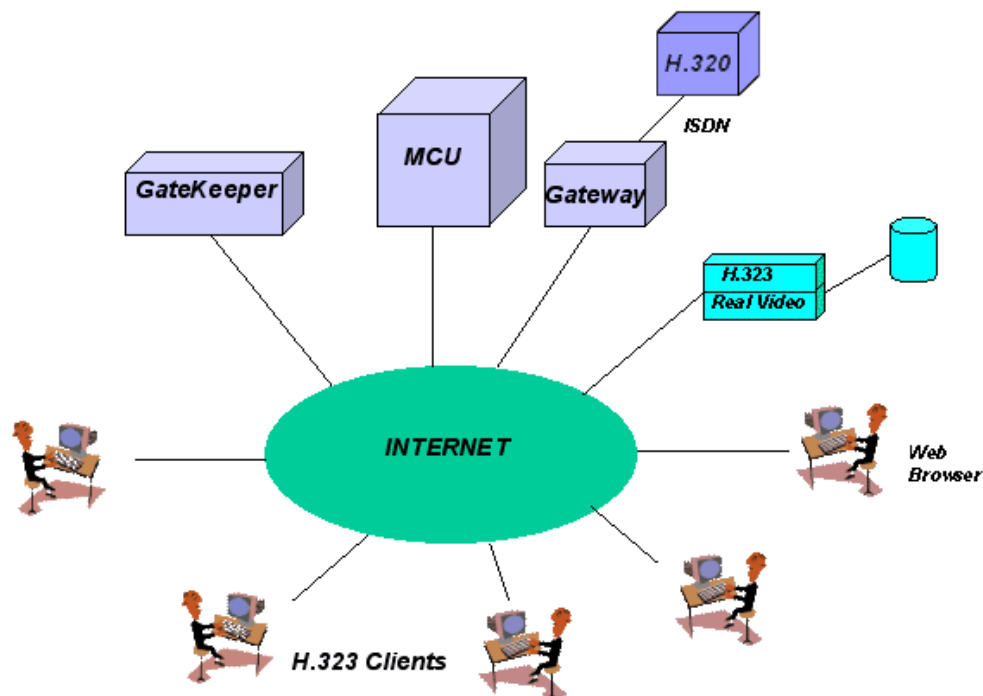


Fig. 7 How the Gateway functions

In addition to supporting pure H.320 videoconferences using H.320 MCUs, gateways between the two protocols can be provided.

In the above picture A user with a standard telephone dials the ISDN telephone number and is connected to the gateway. The telephone user then enters a series of digits to indicate the service/password combination of the desired videoconferencing session. The user can then hear the entire audio from the videoconference and can also interact with others in the conference. The gateway is able to simultaneously connect multiple telephone calls and can even connect to a telephone bridge that could allow participation by a large number of audio only users.

Bandwidth Considerations

The H.323 client terminals encode the selected audio (usually from a microphone) and video (usually from a camera) inputs. The encoded audio and video are then compressed into a single audio/video stream and sent to the remote end point (another client terminal or a MCU).

Different rates can be selected for the encoding process. As an example, an encoding rate of 384 Kbps might be selected. 64 Kbps is reserved for the audio and 320 Kbps is reserved for the video. The 384 Kbps stream is compressed (redundancy is removed) and sent to the remote end point. Similarly a 384 Kbps stream is received from the remote end point. Thus approximately twice 384 Kbps in bandwidth (less any bandwidth saved because of compression) is required to support the videoconference for this end point. If there is a lot of motion in the video, very little compression is achieved. If there is almost no motion in the video, the savings approaches about 50%. Since we must design for the worst case, assume a bandwidth requirement of twice 384 Kbps.

Faster encoding rates can be selected. Most client terminals support rates up to 768 Kbps. Some proprietary implementations can encode at speeds up to 2 Mbps. The higher the encoding rate, the better the quality of the video. However, higher encoding rates also mean higher bandwidth requirements, greater impact on the network, and greater impact on the MCU capacity. Lower encoding speeds can also be selected down to about 128 Kbps. This of course means lower video quality. 384 Kbps is a good compromise between quality on one hand and resource impact on the other. 384 Kbps will support 30 frames per second video. Lower encoding speeds yields lower frame rates and choppy video. There is a discernable but small improvement in quality between 384 Kbps and 768 Kbps.

Chapter II

The VC System

Topics Covered

- Videoconferencing Solutions
- Videoconferencing systems
- Components of the VC system
 - Equipments
 - Client-Endpoint devices
 - NMC-network Management Center equipments

Videoconferencing Solutions

Videoconferencing Solutions can Be Organized Into 3 Main Categories:

1 – Integrated Systems

This is the solution of choice for fixed corporate, educational, and government environments. All equipment is permanently installed, providing the highest quality and ease of use. The equipment runs over a corporate LAN or can utilize ISDN or IP networks. These systems are appropriate for formal rooms and/or to facilitate larger groups.

2 – Roll-Abouts

The hardware used for roll-about solutions is housed on a mobile cart, which allows videoconferencing in multiple rooms using the same system. While the operating quality of the system remains intact, variables such as lighting, sound quality, and ease of dial up/network access can suffer as the equipment moves between various locations.

Roll-abouts are appropriate for small, less formal rooms where the need for increased flexibility and mobility takes precedence over image quality.

3 – Desktop Solutions

These are designed for a single user and are typically installed on top of a computer monitor. Utilizing IP and/or LAN connectivity, these are an effective, reliable and affordable way to begin utilizing videoconferencing or to implement it widely for satellite offices or telecommuters. Image and audio quality meet acceptable business standards depending upon LAN traffic.

The Videoconferencing System

While Choosing a Videoconferencing System, the following are considerations that worth great attention.

Simplicity

- All systems share the same intuitive interface and features
- When you know how to use one component of the VC system, make sure you know how to use them all

Software Platform

- Simplified installation, upgrades and expansion for a strong return on investment

Audio and Video

- Making sure that the Audio Module that delivers unparalleled sound quality
- Making sure that the Camera ensures all participants are seen clearly
- The Call Management- its capability of delivering optimal audio and video without manual intervention
- Systems filter out unwanted background noise such as air conditioners and PCs
- Images are transmitted in their native resolution

Reliability

- System automatically adjustments so that meetings continue uninterrupted

Robust Security

- Highest level of standards-based embedded encryption
- Automatically activated to provide secure conference calls
- No need to purchase additional hardware

Standards-based

- Compatibility with systems from multiple vendors
- Seamless integration with network features

8 Steps to be followed in implementing a videoconferencing system

1) Intended use for your system?

Generally speaking, how do you wish to use the equipment and whom are you going to talk to?

2) Number of sites?

How many offices do you need to communicate with and what resources will each have at their disposal?

3) Number of participants per site?

How many people do you want to participate in video calls at your various locations? Do you want the same videoconferencing set-up for every location? Or, do you want to have a more deluxe system for your headquarters?

4) Size of your room(s)?

Where do you want to put the equipment? For example, is your current conference or meeting room big enough to add a system? Does your current meeting environment offer sufficient size, lighting and ancillary resources to accommodate an appropriate videoconferencing solution?

5) Connectivity?

ISDN, BRI, PRI, T-1, Fractional T, ATM, Frame Relay, xDSL, Cable-modem? What connectivity best suits your internal communication requirements? Will the connection be dedicated to the conferencing system? What types of communications do you plan? Voice, video, data? (The type of communication will determine bandwidth requirements.) Also, what connectivity will remote sites make use of?

6) What type of systems or formats will you be calling?

An ISDN system typically sends and receives voice and video data using the h.320 standard. An IP based (network) system typically communicates using the h.323 voice/video standard. You can only call between systems that subscribe to the same standard. However, gateway and bridge devices are available that will allow you to translate calls between the common standards. (Standards are defined by the International Telecommunications Union and are the same worldwide.)

7) Do you need data capability in addition to video & audio?

Are you just looking to only see and talk to the other people, or are you going to require the ability to do collaborative computing and share data? Do you want to share computer files and documents on screen? Will you need to show PowerPoint presentations?

8) Contact – your vendor Directly!

Once you understand your needs, most vendors will help you in custom design a videoconferencing system that is perfect for your video communication requirements.

Videoconferencing Equipments.

Client-Endpoint devices

Camera

The main camera is critical to the overall quality of the system. In many cases a single camera is installed to capture video at each end of the call, while integrated systems may feature multiple cameras in order to provide a greater range of viewing angles. Features such as zoom; pan and tilt add functionality that can enhance the overall user experience.



Variables such as the size of the room, number of users, seating plan an expected usage should all be considered when specifying a camera.

Audio Components:

Microphones, Mixers and Speakers



Seeing quality video is important for a meeting, but hearing the proceedings is essential. Most expert designers consider High quality audio as important or more important than video. You need a microphone for capturing local audio and speakers for playing the sound received from the far end. Features such as noise suppression and echo cancellation can make an appreciable difference in the perceived sound quality and should be evaluated as part of the system design process.



Codec



The Codec is the heart of the videoconferencing system. The word itself is shorthand for "Coder/Decoder" and describes the function of the equipment.

Audio and video data create enormous amounts of digital data, and without compression simply could not be delivered over digital networks. The codec takes the audio and video captured at your site and compresses the digital data so that it can be transmitted efficiently; when the data arrives at the other end of your conference it is received and decompressed for viewing.

Display

The video being received from the far end is viewed on a display monitor, which can be a computer screen, plasma, a projector display or a TV. While factors such as screen size, brightness and resolution all play a role in the quality of the image, the performance of the codec and the amount of bandwidth available have a more direct affect on image quality.



Document camera

A document camera (doc cam) is essentially an overhead projector with a camera mounted on top. It has lights on each side that allow objects to be illuminated and projected (in high resolution - 1280x1024 pixels) onto the screen using the room's projection system. Controls on the front include freeze, zoom in and zoom out buttons. In addition to displaying traditional transparencies, this device can display any three dimensional object.



Stands & Equipment Carts

Your room aesthetics are usually improved by having all your equipment neatly organized in one spot. Typically, you would place your TV monitor, codec and camera on top of the stand and below you have a cabinet to hold other equipment, like a computer or VCR, etc.

Network Connection

This is the connection that carries data between video systems communicating with one another. The size of the connection and the ability to access it in a consistent manner, determines both video performance and quality of service. The connection can range from an ISDN phone line to a dedicated PRI/T-1 connection or access to a local area network.

When making your connectivity selection, consider the following items to help make the best decision for you:

- Intended Use – Who will you call, using what standard and at what speed? Who will be calling you, at what standard and at what speed?
- Quality of Service – What quality you require, expect and receive. In most cases you only have control over half of this equation. The combination of connectivity and equipment on both ends of the conference add up to your QOS potential.
- Versatility – Will you need to communicate point to point or multi-point, and in what environment IP or ISDN?

User Interface

The user interface is what brings the system to life for users. It allows people to dial calls, control and adjust the various components of the system, and launch other presentation devices (such as a VCR).

Creating an effective user interface requires mastery of the various hardware and components of the system as well as a thorough knowledge of the "culture of use" within the environment, such as organization or campus.

NMC-network Management Center equipments

Video Conferencing Media Manager

Bandwidth management, reporting/billing, scheduling, firewall traversal and security are just a few of the capabilities of the conference management solutions.

A Management server is an Integrated Media Server comprising the following key elements

- Centralized Management and Administration
- Video Telephony / Video PBX
- Advanced Rich Media Applications

For instance With **VCON's Media Exchange manager (MXM)** network administrators have an elegant solution with which to manage their rich media network and users have video telephony features such as call-forward, call-transfer, and ad-hoc conferencing. Whether the network is populated exclusively with VCON product or the network is multi-vendor, the MXM is able to manage and administer services to all videoconferencing systems and devices. Never before has a video network been so easy to manage and has mass deployment of videoconferencing in a large organization been so attainable.



Fig.8 VCON MXM and terminals configurations

Multipoint Control Unit (MCU)

MCU is a device that links three or more videoconference locations for fully interactive audio, data and video teleconferencing.

An ITU-T MCU must meet the requirements of H.323, H.231 and H.243. In addition, the T.12x and T.13x suites of standards, desirable for both codec and MCU, provide greater functionality for multipoint videoconferencing

The MCU enables voice only and multimedia conference calls between H.323 entities and non-H.323 entities. Non-H.323 devices such as telephones or H.320 terminals can join the conference via a gateway.



Fig.9 Radvision's MCU



Fig.10 Multipoint International sessions using MCU

Gateways

A gateway provides a path between H.320 and H.323 systems. It translates H.320 commands and audio/video streams to H.323 audio/video streams and vice versa. Users with H.320 client terminals dial the gateway over ISDN lines. The H.320 client then needs to input a service/password combination for the selected session, and the gateway connects the H.320 terminal to the selected session. All H.323-based users can see and hear the H.320-based users as if they were on H.323 terminals, and similarly the H.320-based users can see and hear the H.323 users as if they were on H.320-based terminals. Multiple H.320 connections can be made to the gateway up to the capacity of the gateway.



For example, The Radvision gateway called ViaIP allows H.320-based videoconferencing client terminals that use ISDN telecommunication lines rather than the Internet for connectivity to join a

H.323, Internet-based videoconference. The gateway also allows users to make telephone calls into the gateway and participate as an audio only user in the videoconference. The ViaIP gateway is connected to a single ISDN PRI line (telephone number 847 467 0001); a second PRI can be added. The capacity of the ISDN PRI line is such that it can simultaneously handle three H.320 terminals running at 384 Kbps plus five telephone users. While the gateway does have the capability to allow H.323 users to make ISDN calls to H.320 sites, we currently have this capability disabled.

Data Collaboration Server (DCS)

This server enables data sharing using the H.323 data encoding standard T.120. The capability to support T.120 calls is handled via Data Collaboration Servers (DCS).

Some MCU sessions (such as service 70) are configured to support DCS. Client terminals that connect to these MCU sessions and are capable of supporting T.120 can share applications, a white board, and text messaging through the videoconference. This capability

does not use the video streams, but instead uses a separate IP path through the network. If the client terminal is a VCON Falcon IP unit, the data sharing will occur using IP Nexus on the same PC that runs the IP Nexus client. If the client terminal is a Polycom Viewstation, the Viewstation will be configured with the IP address of a PC that must run Netmeeting.

Recorders, converters and live video to streaming media servers.

These servers typically run Red Hat Linux and a combination of third-party licensed and written applications for recording, streaming and delivering videoconferencing sessions. The server contains built-in Gigabit Ethernet connections, and has options for Networked File System-mounted storage devices.



Fig.11 Starbak Server

Example:

Starbak's Torrent videoconferencing gateway converts a videoconference into **streaming media**. This would let a manager, for example, make an announcement by videoconference, while others not on the call could watch it later.

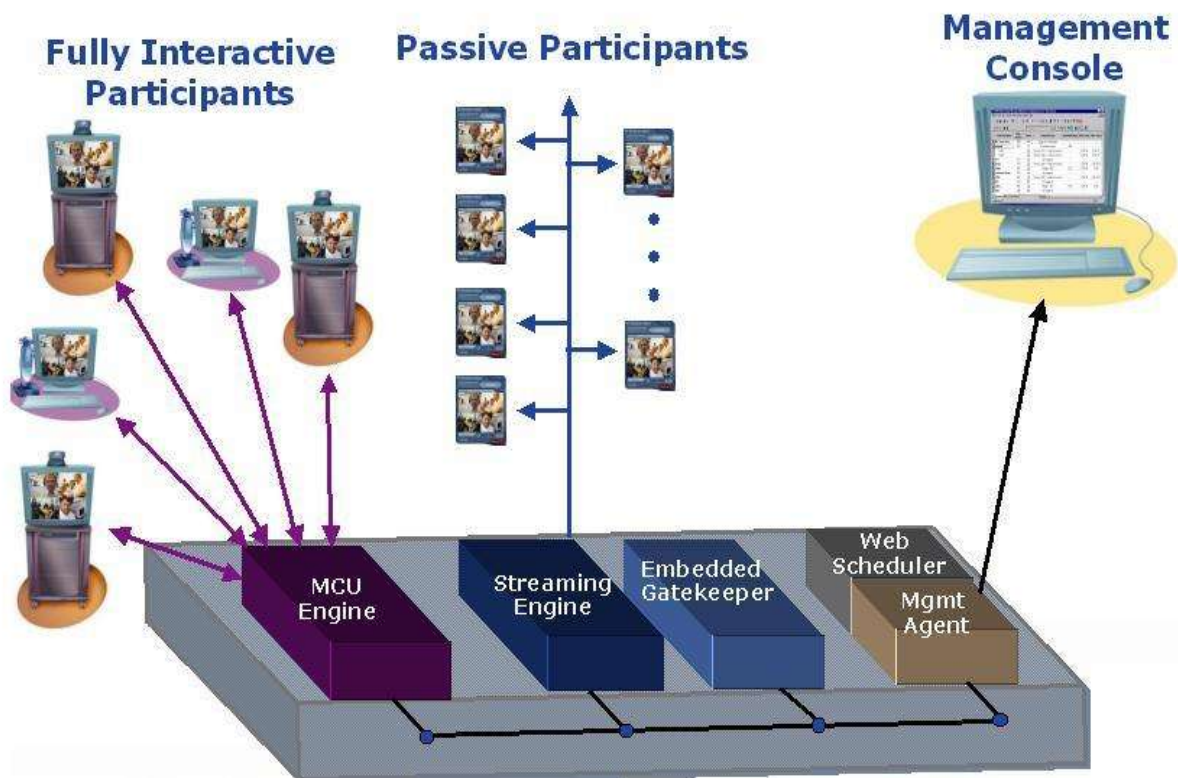


Fig.12 Videoconferencing Servers modeled in VCON's VCB 2000 All-in one multipoint and streaming solution

Appendix A.

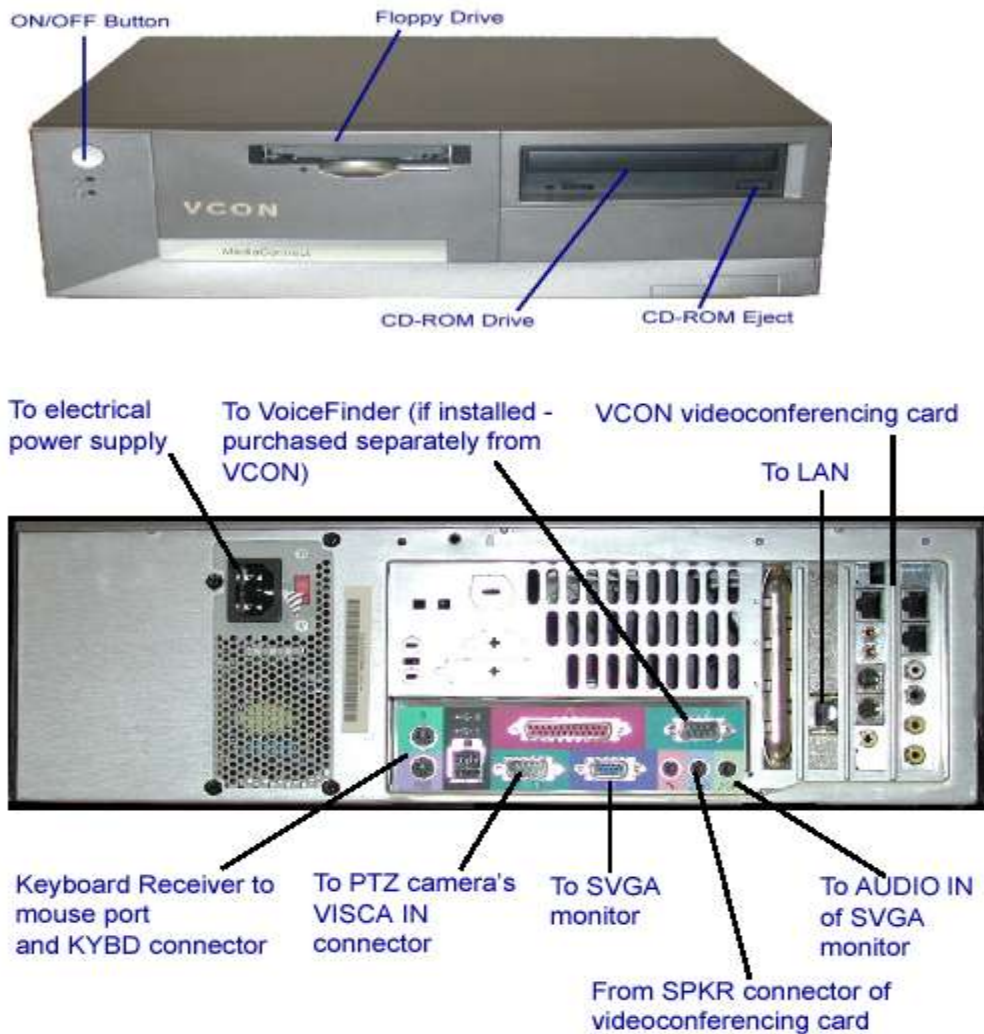
Videoconferencing terminal Examples

I -VCON MC 9000 Sample Installation

Media Connect 9000 System Components

The standard Media Connect 9000 components are:

- MC 900 Computer unit with installed videoconferencing board



- VCON's Pan/Tilt/Zoom (PTZ) camera



- Infrared wireless keyboard and receiver



- Tabletop microphone

- Cart and SVGA monitor with MediaConnect 9000Pro



A range of easy-to-use additional accessories can also be provided to enhance the abilities of Media Connect 9000 such as:

- VCON Voice Finder
- Document camera
- Second PTZ camera
- TV monitor

II- FALCON IP Sample Installations

Falcon IP System Components

Falcon IP is a complete set-top videoconferencing appliance. All the components

Falcon IP Components

Component Model 1 & Model 3

- Falcon IP Main Unit .
- Remote Control .
- Tabletop Microphone .
- Communication cable (LAN/ISDN) CAB42001 . (4)
- Serial DTE (Lap Link) cable CAB42003 .
- TV S-Video cable CAB72008 .
- TV Composite cable CAB72009 .
- Camera S-Video cable CAB90051 .
- Camera Composite cable CAB72010 .
- LAN Cross cable CAB42005 Optional
- Falcon IP User™s Guide .
- Quick Reference Dialing Card .



Fig. Falcon IP Main Unit

➤ **To connect Falcon IP for data sharing**

- ❑ Connect a serial DTE cable (CAB42003) between the Data port of the Falcon IP and the COM1 (or other) port of the computer.

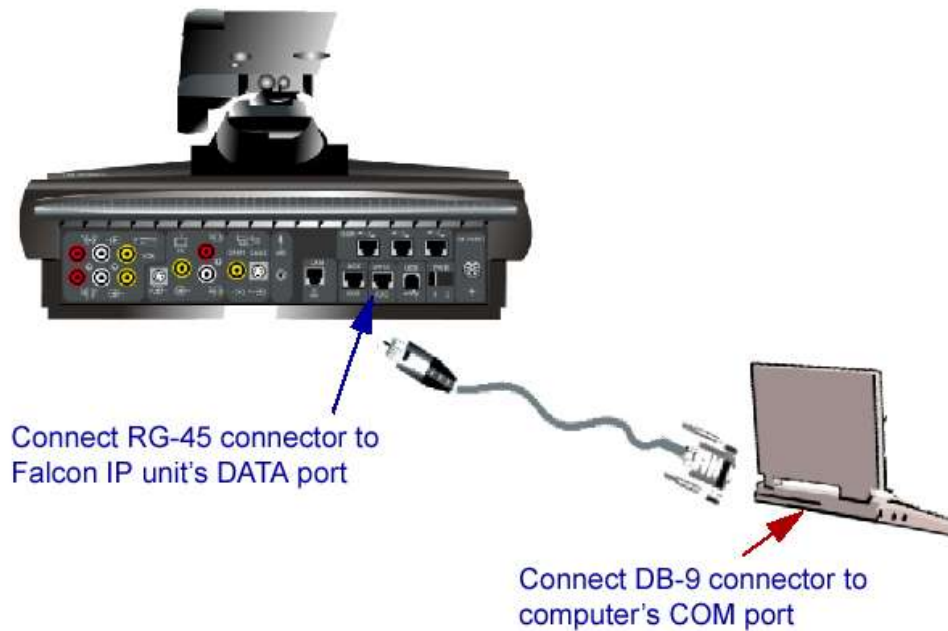
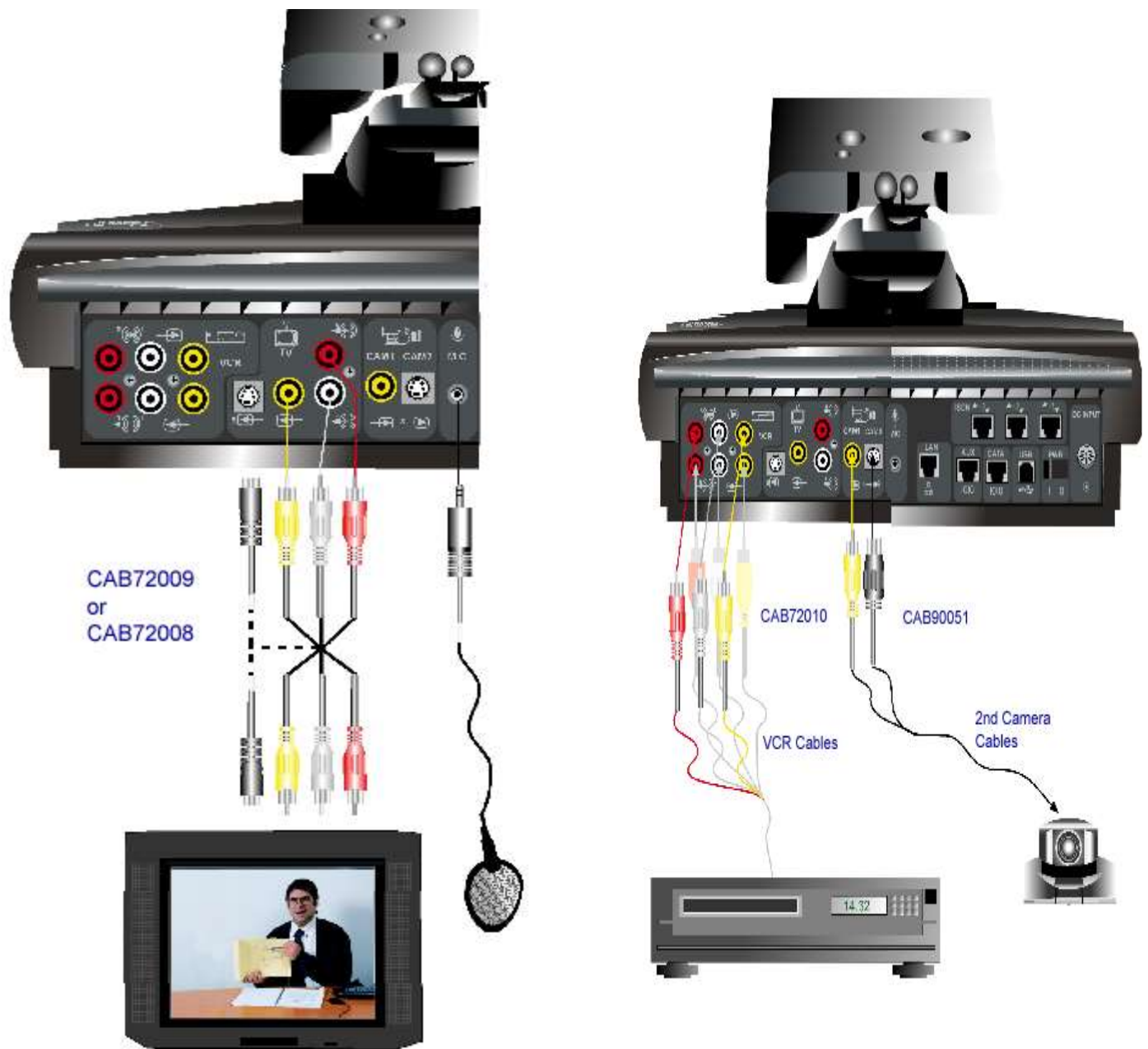


Fig. Remote Control and Its use in configuring Falcon.



Falcon IP Configuration Example:



Appendix B- Glossary of Terms

Analog signals

Audio/video signals currently used in broadcasting where the signal is represented by variable measurable physical quantities (such as voltage). Current TV and radio signals are analog, as are many telephone lines. (Contrast with digital)

Bandwidth

In **casual use**, the amount of information that can be transmitted in an information channel. High bandwidth Internet access means that web graphics load quickly on Netscape. High bandwidth videoconferencing means that the picture and sound will be clear.

In **computers**, the speed at which data can be transmitted on a communications frequency.

In **telecommunications**, the maximum frequency (spectrum) measured in Hertz or cycles per second, between the two limiting frequencies of a channel.

bps

bits per second (lower case is significant)

Broadband

A high-capacity communications circuit/path. It usually implies a speed greater than 1.544Mbps. (Contrast with wideband and narrowband)

Codec

Coder-Decoder. Videoconferencing hardware that codes the outgoing video and audio signals and decodes the incoming signals. Prior to transmission, the codec converts analog signals to digital signals and compresses the digital signals. Incoming audio and video must be decompressed and converted from digital back to analog.

Compressed video

When the vast amount of information in a normal tv transmission is squeezed into a fraction of its former bandwidth by a codec, the resulting compressed video can be transmitted more economically over a smaller carrier. Some information is sacrificed in the process, which may result in diminished picture and sound quality.

Desktop videoconferencing

Videoconferencing on a personal computer. Most appropriate for small groups or individuals (compare with room-based videoconferencing). Many desktop videoconferencing systems support document sharing.

Echo-cancellation

Process of eliminating acoustic echo in a videoconferencing room.

Frame rate

Frequency in which video frames are displayed on a monitor, typically described in frames-per-second (fps). Higher frame rates improve the appearance of video motion. Broadcast TV (full motion video) is 30 frames-per-second.

Gatekeeper: In the H.323 world, the gatekeeper provides several important functions. First, it controls access to the network, allowing or denying calls and controlling the bandwidth of a call. Second, it helps with address resolution, making possible email type names for end users, and converting those into the appropriate network addresses. They also handle all tracking and billing, call signaling, and the management of gateways.

H.261: ITU standard for video coding for videoconferencing. H.261 is a discrete cosine transform (DCT) based algorithm for video in the 64kb/s to 2mb/s range. All H.323 compliant videoconferencing system are required to support this codec.

H.263: ITU standard for video coding within videoconferencing. H.263 offers better compression than H.261, particularly in the low bit rate range used by modem.

H.320: ITU standard for videoconferencing over ISDN and fractional T1 lines.

H.323: ITU standard for videoconferencing over networks that do not guarantee bandwidth, such as the Internet. H.323 is the standard that this cookbook is recommending that most users in the education community should be using.

IP Address: A 4-byte number uniquely defining each unit on the Internet. Forms in conjunction with the Transfer control Protocol (TCP) the TCP/IP

MCU:

In videoconferencing vernacular, a bridge connects three or more conference sites so that they can simultaneously communicate. Bridges are often called MCU's - multipoint conferencing units. In IEEE 802 parlance, a bridge is a device that interconnects LAN's or LAN segments at the data-link layer of the OSI model to extend the LAN environment physically. They work with frames (as opposed to packets) of data, forwarding them between networks. They learn station addresses and they resolve problems with loops in the topology by participating in the spanning tree algorithm. Finally, the term bridge can be used in audio conferencing to refer to a device that connects multiple (more than two) voice calls so that all participants can hear and be heard.

MPEG:

MPEG (Moving Picture Experts Group) is a series of ISO Standards for digital video and audio, designed for different uses and data rates.

Multipoint videoconference

Videoconference with more than two sites. The sites must connect via a video bridge. (Compare with point-to-point videoconference.)

Packet: A unit of information sent across a (packet-switched) network. A packet generally contains the destination address as well as the data to be sent.

Point-to-point videoconference

Videoconference between two sites. (Compare with multipoint videoconference.)

Real-Time: The processing of information that returns a result so rapidly that the interaction appears to be instantaneous. Telephone calls and videoconferencing are examples of real-time applications. These kinds of real-time information not only need to be processed almost instantaneously, but also it needs to arrive in the exact order it's sent. A delay between parts of a word, or the transmission of video frames out of sequence, makes the communication unintelligible. The telephone network is designed for real-time communication

room-based videoconferencing

Videoconferencing using a sophisticated system. Appropriate for large groups (compare to desktop videoconferencing).

video bridge

Computerized switching system which allows multipoint videoconferencing.