COMMONWEALTH OF PENNSYLVANIA
DEPARTMENT OF HUMAN SERVICES

INFORMATION TECHNOLOGY STANDARD

Name Of Standard: **Video Conferencing Standards**	Number: **STD-ENSS013**
Domain: **Network**	Category: **Video Conferencing**
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Abstract:

Videoconferencing in its most basic form is the transmission of synchronized image (video) and speech (audio) back and forth between two or more physically separate locations, simulating an exchange as if the two (or more) participants were in the same physical conversation. This is accomplished through the use of cameras (to capture and send video from a local endpoint), video displays (to display video received from remote endpoints), microphones (to capture and send audio from a local endpoint), and speakers (to play audio received from remote endpoints).

Many state agencies and other business partners of DHS are using video conferencing technology today. The cost of equipment has been reduced significantly over the past few years. In addition, the adoption of the International Telecommunications Union (ITU) Multimedia Teleconferencing Standards make it possible for the interoperability of products and services between disparate vendors

General:

One of the simplest yet most popular uses of videoconferencing is to facilitate attendance at meetings. For meetings that already regularly take place and require face-to-face communication, videoconferencing can substitute for the actual physical presence of remote participants. This reduces travel costs as well as travel time and makes meeting attendance more convenient. It can also make meetings more likely to occur. Frequent and/or ad hoc meetings that might not have been scheduled due to travel costs and timing can be enabled via videoconferencing and enhance the sense of teamwork among people at different locations but working on the same project. Videoconferencing provides remote participants with much of the face-to-face familiarity that comes with physical presence, including elements of facial expression, body language, and eye contact. If videoconferencing is readily available on individual desktops, the cohesive effects of this enhanced communication can be even greater. Collaborative work can then be enhanced further through the integration of videoconferencing with collaborative electronic tools (data transfer, shared whiteboards, and shared applications.)

The quality of the audio and video are critical to the success of the remote participation. Both will affect whether or not the remote participant(s) feel like they are truly part of a meeting (not just an observer) and also whether or not the other participants treat them as part of the meeting. For meetings, though it may seem a bit counter-intuitive, audio is probably more of a "show stopper" than video. Minor hiccups in the video (Pixelization, freezes, etc.) are often tolerated by users. Similar hiccups in the audio make a meeting almost worthless. Therefore audio quality should be paramount. In the specific case of a multi-point meeting -- where more than one location is participating remotely, several factors affect the success of the remote participation. These include the view participants have of each other, how well participants can hear each other and be heard by each other, and how participants determine who is leading the meeting or "has the floor" at any given time.

Video may be either voice activated or continuous presence. Voice activated is where the incoming video from the current speaker’s location is displayed to all other sites. Continuous presence is where each location can see all other locations at the same time using multiple sessions on the display.

Audio may be half or full duplex. Half duplex is likened to “walkie-talkies”, where participants can only hear one speaker at a time and must indicate somehow when speaker control should be passed. Full duplex is “natural” in the sense that everyone can hear everyone else at all times.

Another aspect to consider is how the meeting will be controlled. Under full-duplex audio where people can virtually talk over each other there is very little or no control. **Chair control is a feature that will allow** control to be passed via some designated mechanism such as “electronic hand raising”. The site possessing chair control is seen and heard by others until chair control is passed.

As with any new technology, successful integration of videoconferencing into existing activities requires attention to the needs of the people who will be using it. The determination of what is acceptable and useful must be based on the reaction and comfort level of the end users. In the case of simple point-to-point meetings, there is not a lot of new learning required for participants to successfully interact with each other as long as the video and audio quality do not interfere. Care should be taken to ensure that participants feel they can see and hear each other clearly.

Microphones should be of sufficient quality to pick up the speaker's voice naturally (in terms of volume and physical position) and without excessive background noise. Microphones and speakers should be positioned so that they do not cause feedback and interference with each other, such as when the microphone picks up the sound from the speakers. Using directional microphones will also help limit the interference. Camera quality should be good enough to capture an acceptable image (test with users at the remote site to see how you are coming through) and cameras should ideally be auto-focusing and should auto-adjust for lighting conditions so that participants do not need to adjust them while conferencing. Speaker volume and camera position should be user-adjustable, or have proven acceptable auto-adjusting ability with user override capability. Displays for incoming video should be positioned as naturally and comfortably as possible for inclusion in the meeting and to enable/encourage eye contact. Conference controls that do not duplicate natural conditions (i.e., voice activation in multi-point conferences) may require some training on the part of users to become comfortable and effective in their use.

Attention paid to the total "look and feel" of the meeting scenario prior to conferencing helps to ensure that the technology will enhance rather than detract from the success of the meeting.

Collaboration is the process of working together. Videoconferencing systems can be designed to support rich multimodal interactions between sites. A videoconferencing terminal will generally come with a number of software tools including electronic whiteboards, ftp, and chats. The whiteboard can be useful for dynamic lectures, collaborative diagramming, brainstorming, and sharing notes. Ftp can be used to transfer files quickly without the need for a separate operating system window. Chat can be useful when audio quality is poor or unavailable for some participants or when a subset of participants needs to communicate privately.

An interface is often provided to enable sharing of third party applications that may be installed on participating workstations, particularly useful when group work is supported by project-specific software applications.

Basic Components

Introduction

A videoconferencing terminal must have a few basic components to "get the job done": a camera (to capture local video), a video display (to display remote video), a microphone (to capture local audio), and speakers (to play remote audio). In addition to these more obvious components, a videoconferencing terminal also includes a codec ("COmpressor/DECompressor"), a user interface, a computer system to run on, and a network connection. Each of these components plays a key role in determining the quality, reliability, and user-friendliness of the videoconferencing experience as well as any given videoconferencing terminal's suitability to particular purposes.

Components

The Camera

At least one video source is typically present at each endpoint of a videoconference session. The most common video source is a single main camera that captures live movement occurring at one end so that it may be sent to the other end in near real-time. The quality of the camera may have a significant impact on the appearance of the video at the receiving end. Cameras vary in terms of features that will affect both their usefulness and their cost, such as the ability to pan, tilt, and zoom, wide angle versus narrow angle lens, manual focus versus auto-focus, manual iris versus auto-iris, auto-tracking, remote control, and/or RS-232 control. Naturally, as features are added, cost goes up.

The Video Display

In addition to capturing local video, a videoconferencing solution must include the ability to display the remote video that is being received. This incoming video is displayed on a monitor, most often a computer monitor, which influences how clearly the remote site can be seen and also how many people at the receiving site can easily see it. "Typical" display monitor quality considerations such as screen size and resolution affect the size and clarity of the incoming video window and also the integration of the incoming video window with the application interface that surrounds it. The quality of the image within the video window itself is, however, more directly related to the performance and capabilities of the codec and to the quality and bandwidth of the network connection. Video resolutions supported by H.323 are CIF (352 X 288 pixels) and QCIF (176 by 144 pixels). Since these resolutions are fixed, increasing the network bandwidth beyond a certain point will not show an appreciable difference in video quality within any given video frame. However, additional bandwidth enables higher frame rates (i.e., the sending of additional video frames per second), which can have dramatic improvements on the smoothness and video quality of motion.

Audio Components

Within a videoconference audio is as important, and often considered more important, than video. As long as audio remains intact, loss of video or poor video quality will not impede many of the communication objectives of a conference. In contrast, poor or disrupted audio quality effectively shuts down a videoconference. Participants are left scrambling to find a "native audio" telephone to complete the meeting. For this reason, devices that capture local audio (microphones) and those that reproduce remote audio (speakers) are critical conference components. Coupled with this are characteristics associated with comprehensible full duplex (simultaneous two-way) transmission of audio, such as echo cancellation, noise suppression, and audio mixing. These features are influenced by a combination of the microphones, speakers, and codecs.

The Codec

The codec actually forms the heart of any videoconferencing terminal and is the main enabler of wide-scale videoconferencing. The word "codec" is a shortened version of "compressor/de-compressor" and is specifically applied to the wide variety of algorithms used for actually compressing or decompressing audio and/or video information. This compression has historically been necessary to make the audio/video data "small enough" to be practical for sending over expensive network connections. In this sense, there are many audio and video "codecs" (particular compression/decompression methodologies) that are supported as part of the ITU H.323 Multimedia Teleconferencing Standard. The amount of data required to "describe" audio and video in a digital format is very large by today's data networking standards. Without some form of codec, the transmission of a videoconference requires extremely high amounts of network bandwidth. It is the codec that takes the sights and sounds captured by the local camera and microphone, and then compresses that information such that it may be transmitted across a network fast enough to enable near real-time communication. When the compressed information is received at the remote site, the codec within the remote site's videoconferencing terminal decompresses it and enables "play back" through the speakers and display.

The User Interface

Often consideration and comparison of videoconferencing systems are based solely on video and audio quality -- what it looks and feels like during a conference session. Other features of the system should be considered, such as how to get into and out of conferences, what else can be done in conjunction with a videoconference, and what is known about how the call is going or documentation of the session once the call is over.

Additionally:

- **How the video terminal application "works and plays" with others?** Is the system easy to install, de-install, etc. How much system capacity does the videoconferencing application use? Can other applications run comfortably and reliably when the videoconferencing application is running and in use? Is a wide range of system performance acceptable, or are system requirements stringent? Has the videoconferencing application been tested for interoperability with other H.323 devices?
- **The "Dial" menu, or placing and receiving calls.** Is there an easy to access Phonebook for keeping track of frequently called numbers in a user-friendly way? Is there an automatic call log available for call history and/or error tracking? Can the data rate (call bandwidth) be selected for particular calls in a way that is easily understood?

- **Application sharing and data collaboration.** Are these features fully integrated into the videoconferencing application or are they provided using a "helper" application (e.g., NetMeeting) or perhaps not available at all?

- **Interaction with audio/videoconferencing devices.** Can a wide variety of audio and video devices be used with the terminal application or are only certain devices supported? Are inputs and outputs other than cameras and monitors supported (e.g., VCR in or out?) To what degree can audio/video features (e.g., volume, echo, color, brightness) be controlled from within the application? Is there support for the use of alternate or enhanced devices (e.g., Far End Camera Control, dual monitors, and telephone handsets for privacy?)
- **Support for the ITU H.323 Multimedia Teleconferencing Standard.** How compliant is the video terminal with the current H.323 standard? How prepared is the terminal/developer/vendor to support future H.323 versions and directions? Does the video terminal make any concessions now to cover potential functionality gaps in the current H.323 standard? (e.g. user authentication, secure gatekeeper registration?)

The Supporting System and the Network Connection

Though the supporting system and the network connection are not technically part of the basic components of a videoconferencing terminal, they have a definite effect on the terminal's perceived performance.

Originally deployed over networks that could provide some guarantees about the level of service that would be delivered to the application, ISDN and/or dedicated T1 circuits of the H.320 standards-based world provided predictable delays over somewhat expensive dedicated circuits. H.323 standards-based videoconferencing was engineered to take place on a data network, such as the Internet, without any quality-of-service standard. Such networks were not originally intended for delivery of sensitive near real-time applications. The audio/video data within a videoconference is segmented into chunks by the application, encoded and compressed, put into a series of data packets and sent over the network to the remote end at basically constant intervals. The data packets may arrive at their destination at slightly varying times, if at all, and possibly out of order. To keep the "real time" impression of an interactive videoconference, the packets must arrive, on time and in time to be re-ordered for delivery through the videoconferencing terminal.

When dealing with networks such as the Internet there are five fundamental network problems that must be considered. They are *bandwidth, packet loss, latency, jitter and policies.*

- **Bandwidth** is the fundamental requirement that there be enough space in a network path for all packets to get through unimpeded. For a rough idea of scale, a typical ISDN videoconference uses around 128-384kb/s. IP-based H.323 video systems can use the same bandwidth, although generally they tend to go higher since the network is cheaper, so bandwidth of around 384-768kb/s is very common. Higher-quality videoconferences can go to 1.5-2.0Mb/s, and if broadcast quality is desired the sky is the limit — 6Mb/s for NTSC/PAL transmission, 20Mb/s for prerecorded HDTV and higher for 'live' content. Bandwidth need is symmetric — each end will transmit and receive this amount of traffic. For a multi-point videoconference keep in mind that the MCU/bridge is seeing all of the streams at the same time, even if it is not forwarding them on. So for an 8-site videoconference running at 384kb/s, every site sends and receives 384kb/s to the MCU, and the MCU receives and forwards 8*384kb/s = 3Mb/s roughly.
- **Packet Loss** is when packets fail to arrive correctly. This can be due to insufficient bandwidth along the path (when congestion occurs, routers will drop packets), or perhaps errors in transmission. Errors occur most commonly on wireless links such as microwave, satellite or local wireless Ethernet. They can however also occur on copper and even fiber links. Packet loss results in effects such as "tiling" within the video window, missing pieces or blank areas within the video window, and/or disruptions in audio.
- **Latency** is the time delay between an event occurring and the remote end seeing it. Latency is introduced both by the encoding and decoding process, and hence depends on the equipment

used, as well as the time it takes packets to traverse the network. Little can be done to change the network latency, on any large scale, beyond getting directly involved with a carrier or a research network. The speed-of-light is a limiting factor especially on satellite networks or international cable links. Excessive latency increases the chances of people "talking over one another" because they don't realize that the person at the other end has started speaking too. Another problem is that the latency for the audio and video may be different, and hence lip movements don’t appear synchronized with the audio. This is a function of both the terminal and the network, and can vary dramatically — some products try to compensate for it.

- **Jitter** is the random variation in latency, due to things like other processes running on the terminal (for example on a desktop PC), other traffic temporarily blocking the path through the routers along the way, or even the network path changing during a videoconference. In extreme cases this results in packets arriving out of order from their transmitted order. Jitter results in uneven and unpredictable quality within a videoconference. Terminals will try to compensate for this by buffering the traffic up to some finite time, before playing it out. This increases latency even further.
- **Policies** are introduced by components like firewalls and network address translation (NAT) devices that are generally used to try to hide or protect network elements from the wider Internet. H.323 typically uses dynamically allocated ports, and is thus not very firewall-friendly. Unfortunately there are very few technical solutions to these, and require you to discuss policy issues with your, or their, network managers.

Nowadays most campus or corporate desktops have 10Mb/s or 100Mb/s Ethernet interfaces to the LAN, so a videoconference is not much of an impact. However, a LAN needs to support "switched" Ethernet, rather than "shared", to be sure the videoconference does not impact others. A "shared" LAN divides the bandwidth between everybody, so on average less bandwidth is available and the risk of packet loss and jitter increases. In some cases, older wiring, such as that installed at many university campuses, may have to be upgraded. Category-5 (or better) horizontal network wiring combined with fiber optic vertical wiring is recommended.

There are additional features offered by Ethernet switches that may prove of value. One is support for IP multicast and another is 802.1Q traffic prioritization. IP multicast is a bandwidth efficient way of delivering data, such as video and voice, to multiple recipients using a single copy for all rather than one copy each. 802.1Q traffic prioritization allows for some devices to get a higher priority for network traffic than other devices — sometimes described as "layer 2 quality of service". Across a campus, and out into the wide area is where most bottlenecks occur. Wide-Area links especially tend to be expensive and often narrower than one would like.

The exact path along the network between video terminals, or from terminals to the MCU, will also affect the performance of a conference. Network packets do not necessarily take the shortest path from one location to another; routers determine which path is taken. A router must examine the destination address of the packet and then calculate where to send it. Every pass through a router is called a "hop". Because a calculation is involved, every "hop" adds a bit of delay to the total time required to transit the entire path, increasing latency and jitter, and also provides bottlenecks (routers have only so much memory and CPU capacity), increasing the risk of packet-loss. The fewer hops along the path the better the result.

Given all of these potential problems it is best to make no assumptions at all about a network's readiness for videoconferencing. It is strongly recommended that discussions of videoconferencing plans should be made with the Department’s network staff early on in the decision process to determine if the network will support videoconferencing in the preferred locations, including off-campus sites. Additionally this staff can advise of potential issues, monitor the network for problems, and possibly engineer a better setup.

Standard:

Authoritative Point of Contacts:

It is assumed that all project teams are familiar with DHS standards and have reasonable access to subject matter experts to resolve issues or to refine approaches. It is further assumed that technical issues are not routinely resolved at high level business meetings. With this in mind any issues involving the use of Video Conferencing should be submitted to the appropriate team or domain leads before elevation to a business issue.

Concerns and issues regarding the utilization of Video Conferencing on the Department's data network should be directed to:

Bureau of Information Systems

Division of Technology Engineering

Enterprise Network & Security Section

Network Architecture and Telecommunications Management Unit

DHS has accepted the industry standard ITU Multimedia Teleconferencing Standards as the de facto guideline for the acquisition, deployment, and operation of video conferencing technology.

The ITU T.120, H.320, H.323, and H.324 standards comprise the core technologies for multimedia teleconferencing.

The T.120 standard covers the document conferencing and application sharing (also known as data conferencing) portion of a multimedia teleconference. The recommendations specify how to efficiently and reliably distribute files and graphical information in real-time during a multipoint multimedia meeting. The objective of the T.120 standard is to assure interoperability between terminals without either participant assuming prior knowledge of the other system; permit data sharing among participants in a multimedia teleconference, including white board image sharing, graphic display information, and image exchange, application sharing, and specify infrastructure protocols for audio graphic or audiovisual applications.

The H.320 standard addresses videoconferencing over circuit switched networks and services like ISDN or Switched-56. This standard 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 protocols that allow a multimedia terminal to utilize the communications links and synchronization of audio and video signals.

The H.323 standard was originally developed as an adaptation of H.320. Corporations have increasingly implemented LANs and LAN gateways to the WAN. H.323 has evolved beyond a logical and necessary extension of the H.320 standard to include corporate intranets and packet-switched networks generally. H.323 utilizes Real-Time Protocol (RTP/RTCP) from the Internet Engineering Task Force (IETF), along with internationally standardized codecs. H.323 is also being used for video and other communications over the Internet.

The H.324 standard addresses and specifies a common method for sharing video, data, and voice simultaneously using v.34 modem connections over a single analog (POTS) telephone line. It also specifies interoperability under these conditions, so that videophones, for example, based on H.324 will be able to connect and conduct a multimedia session. H.324 has the broadest impact in the marketplace because it incorporates the most pervasive communications facility – POTS – installed today on a global basis.

Support for these standards will ensure that users can call, connect, and communicate with people using compatible conferencing products from other companies and can take advantage of conferencing services that also support these standards.

Glossary of Terms

Algorithm

Some videoconferencing systems offer both proprietary and standard compression algorithm. An algorithm is any step-by-step problem-solving procedure. Transmission of compressed video over a communications network requires sophisticated compression algorithms.

Analog Signal

A type of signal that encodes voice, video, or data transmitted over wire or through the air, and is commonly represented as an oscillating wave. An analog signal can take any value in a range and changes smoothly between values, as opposed to digital signals, which is characterized by discrete bits of information in numerical steps. An analog signal can transmit analog or digital data.

ADC

Analog-to-Digital Conversion

Process of converting analog signals to a digital representation. DAC represents the reverse translation.

ATM

Asynchronous Transfer Mode

A high bandwidth, high speed (up to 155 Mbps), controlled-delay fixed-size packet switching and transmission system integrating multiple data types (voice, video, and data). Uses fixed-size packets also known as "cells" (ATM is often referred to as "cell relay").

Asynchronous Transmission

A mode in which the sending and receiving serial hosts know where a character begins and ends because each byte is framed with additional bits, called a start bit and a stop bit. A start bit indicates the beginning of a new character; it is always 0 (zero). A stop bit marks the end of the character. The time interval between characters may be of varying lengths. Synchronous data uses an external reference clock to unify both ends of the data circuit.

Audio

In video communications, electrical signals that carry sounds. The term is also used to describe systems concerned with sound with recording and transmission; speech pickup systems, transmission links that carry sounds, amplifiers and the like.

Autonomous System

Internet (TCP/IP) terminology for a collection of gateways (routers) that fall under one administrative entity and cooperate using a common Interior Gateway Protocol (IGP).

B Channel

A 56Kbps or 64-kbps channel that carries user data on a line using ISDN D-channel signaling.

B-Mac

A method of transmitting and scrambling television signals. In such transmissions MAC (Multiplexed Analog Component) signals are time-multiplexed with a digital burst containing digitized sound, video synchronizing, authorization, and information.

Bandwidth

Bandwidth is the data capacity of a service, measured in thousands of bits per second (kbps) or millions of bits per second (Mbps). In videoconferencing systems a larger bandwidth is used to spread or "dither" the signal in order to prevent interference.

Baseband

The basic direct output signal in an intermediate frequency based obtained directly from a television camera, videoconference television receiver, or video tape recorder. Baseband signals can be viewed only on studio monitors. To display the baseband signal on a conventional television set a "modulator" is required to convert the baseband signal to one of the VHF or UHF television channels, which the television set can be tuned to receive.

Bit Error Rate

The percentage of received bits in error compared to the total number of bits received. A bit error rate of 10-6 means that there is an average of one error per million bits.

bps

bits per second

A unit of measurement of the speed of data transmission and thus of bandwidth. Actually a nested acronym, meaning binary digits per second. (lower case is significant)

Bps (or BPS)

(8-bit) bytes per second (upper case is significant)

BRI

Basic Rate Interface

One of two ISDN subscriber "interfaces" in ISDN. 2 voice (B) channels at 64 kbps channels and 1 data signal (D) channel at 16 kbps. The B-channels are for voice, video, and data. The D-channel is for signaling between telephone company switches and for carrying ISDN user-network messages.

Bridge

In videoconferencing vernacular, a bridge connects three or more conference sites so that they can simultaneously pass data, voice, or video. Videoconferencing bridges are often called MCU’s – multipoint conferencing units. (See router).

Broadband

A way of transmitting large amounts of data, voice, and video that is greater than telephony networks. In ISDN, broadband channels support rates above the primary rate (1.544 Mbps or 2.048 Mbps). (See wideband and narrowband)

Business Television

Corporate communications tool involving video transmissions of information via videoconference. Common uses of business television are for meetings, product introductions and training.

Camera presets

Allows pre-defined camera angles to be programmed into a videoconferencing system.

CCITT

Consultative Committee for international Telegraphy and Telephony

(Now called the International Telecommunications Union's Telecommunications Standardization Sector or TSS.) The world's leading telecommunications standards organization responsible for establishing interoperability standards for communications systems.

CIF

Common Intermediate Format

An international standard for video display formats developed by TSS. The QCIF format, which employs half the CIF spatial resolution in both horizontal and vertical directions, is the mandatory H.261 format. QCIF is used for most desktop videoconferencing applications where head and shoulder pictures are sent from desk to desk.

CODEC

COder-DECoder

In the videoconferencing world, a video codec converts analog video signals from a video camera to digital signals for transmission over digital circuits, and then converts the digital signals back to analog signals for display.

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.

An uncompressed NTSC signal requires about 90 Mbps of throughput, greatly exceeding the speed of all but the fastest and shortest of today's networks. Squeezing the video information can be accomplished by reducing the quality (sending fewer frames in a second or displaying the information in a smaller window) or by eliminating redundancy.

Compression

Compression is a technique that reduces the quantity of bandwidth or bits required to encode a block of information so that it occupies less space on a transmission channel or storage device and a fundamental concept of video communications.

DBS

Direct broadcast videoconference

Refers to a service that uses videoconferences to broadcast multiple channels of television programming directly to home mounted small-dish antennas.

D-Channel

A channel that carries WAN synchronization information on a line using ISDN D-channel signaling.

Delay

The time it takes for a signal to go from the sending station through the videoconference to the receiving station. This transmission delay for a single hop videoconference connection is very close to one-quarter of a second.

Demodulator

A videoconference receiver circuit which extracts or "demodulates" the "wanted" signals from the received carrier.

Desktop videoconferencing

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

Digital Signal

A way of sending voice, video, or data that reconstructs the signals using binary codes (1s and 0s) for transmission through wire, fiber optic cable, videoconference, or over air techniques. Digital audio/video signals represented by discrete variations (in voltage, frequency, amplitude, location, etc.) can be transmitted faster and more accurately than analog signals.

Distance learning

The incorporation of video and audio technologies so that students can "attend" classes and training sessions that are being presented at a remote location. Distance learning systems are usually interactive and are becoming a highly-valuable tool in the delivery of training and education to widely-dispersed students or in instances where the instructor cannot travel to the student's site.

Document sharing

A feature supported by many videoconferencing systems that allows participants of a videoconference to view and edit the same computer document.

Domain Name

In the Internet, a part of a naming hierarchy consisting of a sequence of names separated by periods (dots) that corresponds to the network number in the IP address. In the symbolic name jdoe@state.pa.us, the domain name is state.pa.us.

DTE

Data Terminal Equipment

As defined in the RS-232 specification, equipment to which DCE (Data Communications Equipment) is connected, such as a videoconference terminal, LAN bridge or router.

Dual 56

Two switched 56 calls made between videoconferencing equipment to allow data transfer at 112 kbps. The videoconferencing equipment performs a two-channel inverse-multiplexing procedure to assure channel alignment.

DVB

Digital Video Broadcast

The standard for direct broadcast television in Europe and the U.S. Based on MPEG2 Compression.

Earth Station

The term used to describe the combination or antenna, low-noise amplifier (LNA), down-converter, and receiver electronics used to receive a signal transmitted by a videoconference.

Echo-cancellation

Process which attenuates or eliminates the acoustic echo effect on videoconference calls. Echo cancellers are largely replacing obsolete echo suppressers.

Echo Effect

A time-delayed electronic reflection of a speaker's voice. This is largely eliminated by modern digital echo cancellation.

Echo suppression

To reduce annoying echoes in the audio portion of a videoconference, it silences all sound when on by temporarily deadening the communication link in one direction. Unfortunately, not only the echo is stopped but also the remote end's new speech, which results in clipping.

EIRP

Effective Isotropic Radiated Power

This term describes the strength of the signal leaving the videoconference antenna or the transmitting earth station antenna. The transmit power value in units of dBW is expressed by the product of the transponder output power and the gain of the videoconference transmit antenna.

FCIF/QCIF

Standards-based formats for communicating between videoconferencing systems from different vendors. QCIF is one quarter of the resolution of FCIF.

FDX

Full-duplex
Two-way, simultaneous transmission of data; a communication protocol in which the communications channel can send and receive data at the same time. Compare to half-duplex, where information can only be sent in one direction at a time.

Fractional T1

Service offering data rates between 64 kbps (DS0 rate) and 1.536 Mbps (DS1 rate), in specified intervals of 64 kbps. It is typically provided by a carrier in lieu of a full T-1 connection and is a point-to-point arrangement. A specialized multiplexer is used by the customer to channelize the carrier's signals.

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.

Frame store

A system capable of storing complete frames of video information in digital form. This system is used for television standards conversion, computer applications incorporating graphics, video walls and various video production and editing systems.

Full duplex audio

2-way audio simultaneously transmitted and received without any interference or "clipping." A common feature of room-based videoconferencing systems.

Full-motion video

In the videoconferencing world, the term "full-motion video" is often used and misunderstood. Videoconferencing systems cannot provide 30 fps for all resolutions at all times nor is that rate always needed for a high-quality, satisfying video image. Picture quality must sometimes be sacrificed to achieve interactive visual communication economically. Videoconferencing vendors often use "full-motion video" to refer to any system that isn't still-frame. Most videoconferencing systems today run 10 to 15 fps at 112 Kbps.

Full motion video is equivalent to broadcast television video with a frame rate of 30 fps for NTSC signals or 25 fps for PAL signals. Images are sent in real time and motion is continuous. Also known as continuous-motion video.

Gateway

Gateways are points of entrance to and exit from a communications network. Viewed as a physical entity, a gateway is that node that translates between two otherwise incompatible networks or network segments.

H.320 / H.323

Sets of widely-used CCITT video compression standards describing methods to allow videoconferencing systems from different manufacturers to interoperate. They include a number of individual recommendations for coding, framing, signaling and establishing connections. Three audio algorithms, G.721, G.722 and G.728, are also included in the standards.

Half duplex audio

2-way audio transmitted and received in turn (rather than simultaneously) so only one site can speak at a time. Contrast with full duplex audio.

Handshake

Prior to a videoconferencing transmission, the codecs exchange predetermined electrical signals that allow them to interoperate by they seeking out a common algorithm.

IETF

Internet Engineering Task Force

A large, open, international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet. The IETF is the protocol engineering and development arm of the Internet.

Internet Gateway

A gateway for accessing the Internet, which is loosely defined as the complex of wide area networks (WANs) joining government, university, corporate and private computers (nodes) in a vast web of network interconnection.

IP

Internet Protocol

IP Address

An address that uniquely identifies each host on a network or Internet. An IP address has a length of 32 bits, and is divided into four 8-bit parts, each separated by a period, as in 164.156.57.10. This kind of notation is called dotted decimal notation. Each part can consist of a number between 1 and 255. In addition to an IP address, you can use a symbolic (domain) name provided by Domain Name Services (DNS) to designate an Internet address.

ISDN

Integrated Services Digital Network

An international standard for end-to-end digital transmission of voice, data, and signaling. In a videoconference it is a system that provides simultaneous voice, video, and text transmission between individual desktop videoconferencing systems and group (room) videoconferencing systems.

ITU

International Telecommunications Union

Headquartered in Geneva, Switzerland, an international organization through which governments and the private sector coordinate global telecommunications networks and services. ITU activities include the coordination, development, regulation, and standardization of telecommunications as well as organization of regional and world telecommunications events.

Kbps

Kilobits per second.

Refers to transmission speed of 1,000 bits per second.

LAN

Local Area Network

A network that interconnects devices over a geographically small area, typically in one building or a part of a building. The most popular LAN type is Ethernet. LANs allow the sharing of resources and the exchange of both video and data.

LAN/WAN Connectivity

This is the practical set of tools, from OS layer protocols to support services, that make a remote access device an effective link between LANs and WANs. An effective remote access server must include a host of communications and translation protocols to fulfill this function.

Latency

The minimum time required to move data from one point to another. Once latency is present, it cannot be optimized. The cause has to be removed (as in using an internal device rather than an external one to remove the latency caused by the serial port). To maximize throughput, use the highest bandwidth available.

Leased Lines

A circuit rented for exclusive use twenty-four hours a day, seven days a week from a telephone company. The connection exists between two predetermined points and cannot be switched to other locations.

MAC

Media Access Control

A system of rules used to move data from one physical medium to another.

MAC (A, B, C, D2)

Multiplexed analog component

Color video transmission system. Subtypes refer to the various methods used to transmit audio and data signals.

Margin

The amount of signal in dB by which a satellite videoconference system exceeds the minimum levels required for operation.

MAX

Media Access Exchange

It supports up to 32 host ports or direct Ethernet connections and up to 8 Mbps to the network. It supports multiple applications, including remote LAN access, leased line backup and individual videoconferencing units, as well as connecting videoconference MCUs to the digital dial-up network.

MBONE

Multicast / Multimedia Backbone

A collection of Internet routers that support IP multi-casting. The MBONE is used as a "broadcast" channel on which various public and private audio and video programs are sent.

Mbps

Megabits per second

Modulation

The process of manipulating the frequency or amplitude of a carrier in relation to an incoming video, voice or data signal.

Modulator

A device which modulates a carrier. Modulators are found as components in broadcasting transmitters and in videoconference transponders.

Multiplexing

Techniques that allow a number of simultaneous transmissions over a single circuit.

Multipoint

Communication configuration in which several terminals or stations are connected. Compare to point-to-point, where communication is between two stations only.

MCU

Multipoint Control Unit

Videoconferencing equipment that allows more than three individual videoconference units to connect together to form a multiparty videoconference session. The MCU uses fast switching techniques to patch the presenters or speaker's input to the output ports representing the other participants.

Multipoint Videoconference

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

Narrowband

A low-capacity communications circuit/path. It usually implies a speed of 56Kbps or less. (Contrast with wideband and broadband)

Network

A group of stations (computers, telephones, or other devices) connected by communications facilities for exchanging information. Connection can be permanent, via cable, or temporary, through telephone or other communications links. The transmission medium can be physical (i.e. fiber optic cable) or wireless (i.e. satellite).

NT1

Network Terminator Type 1

The NT-1 is physically connected between the ISDN board of your videoconferencing system and your ISDN phone line and converts the two-wire line coming from your telephone company into a 4-wire line. And provides network maintenance functions such line maintenance access, timing, and echo cancellation. NT1s may be built into other pieces of equipment or stand alone.

NTSC

National Television Standards Committee

United States' standard for scanning television signals that has been adopted by numerous other countries. Frames are displayed at 30 frames per second. (Other standards: PAL (Europe) and SECAM (France/former USSR))

Packets

A block of information sometimes called a cell, frame, data unit, service unit, or signaling unit. Although each of these elements do have unique attributes, in essence, all are packets.

PAL

Phase Alternative Line System

The European TV standard for scanning television signals. Frames are displayed at 25 frames per second. Used in most European countries. (Other standards: NTSC (USA) and SECAM (France/Former USSR))

Point-to-point videoconference

Videoconference between two sites. (See Multipoint videoconference.)

POP

Point of Presence

This is a point-of-presence of an Internet service provider, used to facilitate remote users' access to the range of applications and IP addresses in the internetwork.

PPP

Point-to-Point Protocol

Provides a standard means of encapsulating data packets sent over a single-channel WAN link. It is the standard WAN encapsulation protocol for the interoperability of bridges and routers. PPP is also supported in workstations, allowing direct dial-up access from a personal computer to a corporate LAN or Internet Service Provider. Using PPP ensures basic compatibility with non-Ascend devices. Both the dialing side and the answering side of the link must support PPP.

PRI

Primary Rate Interface

An ISDN subscriber line, consisting of twenty-three 64 kbps B channels in North America (thirty 64 kbps channels elsewhere) and one 64 kbps D channel, used for signaling purposes.

Promiscuous Mode

A Bridging parameter mode that determines that the Ethernet controller in an Ascend unit accepts all packets and passes them up the protocol stack for a higher-level decision on whether to route, bridge, or reject them. This mode is appropriate if you are using an Ascend unit as a bridge.

Proprietary compression algorithm

A vendor-specific algorithm for compression of a video signal. A videoconferencing system using a proprietary algorithm can only communicate with a remote site using the same algorithm. Many vendors also adhere to standard compression algorithms to facilitate communication across platforms. (i.e .H.320)

Public room

Videoconferencing service offered to the public on a fee-for-usage basis.

Px64

Common reference to the CCITT standards (H.261 et. al.) which describe methods to allow for videoconferencing system interoperability.

QoS

Quality of Service

Interactive video conferencing requires a high QoS. QOS is important as it determines the quality of your video call. Low quality of service results in latency and a jerky picture with poor and inconsistent audio quality.

QPSK

Quadrature Phase Shift Keying

System of modulating a videoconference signal.

Real-Time

The processing of information that returns a result so rapidly that the interaction appears to be instantaneous. Videoconferencing is an example of a real-time application. This kind of real-time information not only needs to be processed almost instantaneously, but 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.

Receiver (Rx)

An electronic device which enables a particular videoconference signal to be separated from all others being received by an earth station, and converts the signal format into a format for video, voice or data.

Room (Group) based videoconferencing

Videoconferencing using a sophisticated system. Appropriate for large groups (See Desktop videoconferencing).

Router

A device or setup that finds the best route between any two networks, even if there are several networks to traverse. Like bridges, remote sites can be connected using routers over dedicated or switched lines to create WANs

RS-232-C

A set of EIA standards specifying various electrical and mechanical characteristics for interfaces between computers, terminals, and modems. The standard applies to both synchronous and asynchronous binary data transmission at rates below 64 kbps.

RS-449

An EIA standard for a 37-pin data communications connector, usually used with RS-422 or RS-423 electrical specifications.

SECAM

A color television system developed by the French and used in the former USSR. Secam operates with 625 lines per picture frame and 50 cycles per second. It is incompatible with the European PAL system or the U.S. NTSC system.

Serial Host

A device, such as a videoconferencing codec, that is connected to a serial host port communicating over a point-to-point link. To a serial host, the MAX appears to be a cable or DCE (Data Communications Equipment).

Serial Host Port

The V.35, RS-499, or X.21 port on the MAX.

Serial Host Port Module

A module on the MAX that connects to a serial host through its serial host port.

Standard compression algorithm

An algorithm convention for compression of a video signal. Adherence to standards allows communication among a wide variety of videoconferencing systems, though not with the same clarity as two similar systems using a proprietary algorithm. H.320/H.323 are the most widely accepted standards in use today.

Switched 56

A dial-up network-based service providing a data channel operating at a rate of 56 kbps. Also a type of network access line, used to provide access to switched 56 network services.

SDSAF

Switched Digital Services Applications Forum

A consortium of equipment vendors, service vendors, and users, with the goal of advancing the state of switched digital services.

SVC

Switched Virtual Circuit

A path over a packet-switched network that appears to be a dedicated circuit, but in fact the connection only stays up as long as needed, and then ends.

Synchronization

In serial data transmission, a method of ensuring that the receiving end can recognize characters in the order in which the transmitting end sent them, and can know where one character ends and the next begins. Without synchronization, the receiving end would perceive data simply as a series of binary digits with no relation to one another. Synchronous communication relies on a clocking mechanism to synchronize the signals between the sending and receiving machines. (See Asynchronous Transmission)

T1

In North America, T1 service delivers 1.544 Mbps, whereas ISDN service delivers 128 kbps. The data travels over the line at the same speed, but for T1 lines, the capacity is twelve times that of ISDN. Typically channelized into 24 DS0s, each capable of carrying a single voice conversation or data stream. The European T1 or E1 transmission rate is 2.048 million bits per second.

T3

DS-3
In North America, a digital channel which communicates at 45 Mbps, or 28 T1 lines.

T1 PRI line

A T1 line that uses 23 B channels for user data, and one 64 kbps D channel for ISDN D channel signaling. This type of PRI line is a standard in North America, Japan, and Korea.

TDMA

Time division multiple access

Refers to a form of multiple access where a single carrier is the shared by many users. Signals from earth stations reaching the videoconference consecutively are processed in time segments without overlapping.

Telecommuter

A work-at-home computer user who connects to the corporate LAN backbone using remote access technologies.

Uplink

The earth station used to transmit signals for a satellite videoconference.

V.35

Commonly used to describe electrical characteristics and connector characteristics for a high speed synchronous interface between DTE and DCE. Originally V.35 described a 48 kbps group band modem interface with electrical characteristics defined in an appendix. Although V.35 is considered obsolete and no longer published by the CCITT, its legacy lives on in the data communications world in the form of the electrical characteristics originally described in the appendix.

Video bridge

Computerized switching system which allows multipoint videoconferencing.

Videoconferencing

Communication across long distances with video and audio contact that may also include graphics and data exchange. Digital video transmission systems typically consist of camera, codec (coder-decoder), network access equipment, network, and audio system.

WAN

Wide Area Network

A data network typically extending a LAN outside a building or beyond a campus. Typically created by using bridges or routers to connect geographically separated LANs. WANs include commercial or educational dial-up networks such as CompuServe, InterNet and BITNET.

White boarding

A term used to describe the placement of shared documents on an on-screen "shared notebook" or "whiteboard." Videoconferencing software includes tools that enable you to work with familiar tools to mark up the electronic whiteboard much like you do with a traditional wall mounted board.

Wideband

A medium-capacity communications circuit/path. It usually implies a speed from 64Kbps to 1.544Mbps. (Contrast with broadband and narrowband)

X.21

A set of CCITT specifications for an interface between DTE and DCE for synchronous operation on public data networks. Includes connector, electrical, and dialing specifications.

Exemptions from this Standard:

There will be no exemptions to this standard.

Refresh Schedule:

All standards and referenced documentation identified in this standard will be subject to review and possible revision annually or upon request by the DHS Information Technology Standards Team.

Standard Revision Log:

Change Date	Version	Change Description	Author and Organization
01/22/0203	1.0	Initial creation	Anthony Ruffner
02/15/2005	1.1	Reviewed for content – Revised Dates and organization structure	Doug Rutter
12/31/2007	1.2	Reviewed & placed into new format	Doug Rutter
06/25/2010	1.2	Reviewed	Doug Rutter
02/17/2011	1.2	Reviewed content – No changes	Doug Rutter
11/18/2013	1.3	Reviewed content – Minor Grammar Changes	Matthew Messinger
3/30/2015	1.4	Reviewed content. Changed DPW to DHS.	Bob Gordon, BIS-DTE
03/04/2016	1.5	Updated the CTO's name	Aamir Qureshi, BIS-DTE