TANDBERG MCU and IP

TANDBERG

D50183, Rev 1.1

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1. Introduction

H.323 is an International Telecommunications Union (ITU) standard that describes the protocols, services and equipment necessary for multimedia communications including audio, video and data on networks without guaranteed Quality of Service (QoS). These networks technologies may include Ethernet, Fast Ethernet, and Token Ring and protocols like Internet Protocol (IP) or Integrated Packet Exchange (IPX). Due to the need to communicate between smaller networks connected to the Internet, IP will be a more popular transport for H.323.

Today, the dominant method of Internet communications is email. However, there is a growing need to increase communications to include audio, video and data. The explosion of the Internet in the early 1990's has paved the way to higher bandwidth connections to corporate offices, universities and even to the home. Now that the bandwidth is available, the demand for multimedia communications over the Internet is growing.

2. What is H.323?

H.323 is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for "terminals and other entities that provide multimedia communications services over Packet Based Networks (PBN) which may not provide a guaranteed Quality of Service".

2.1 Components

H.323 specifies several new standards to allow for communications between terminals on IP networks. These standards dictate how different mandatory and optional components of the H.323 standard interoperate with each other. The major network components of H.323 include the mandatory terminal, and the optional gatekeeper, gateway and multipoint control unit (MCU).

2.1.1 Terminals

The terminal or endpoint must support a minimum of G.711 audio, H.225, H.245, Q.931 and RTP. If the terminal supports video, it must support a minimum of H.261 QCIF. The terminal may support T.120 data sharing although this support is optional. The TANDBERG 500, 550, 800, 880, 1000, 2500, 6000, 7000 and 8000 video conferencing systems all fit the definition of a H.323 terminal.

TANDBERG 1000

TANDBERG 880

TANDBERG 8000

2.1.2 Gatekeepers

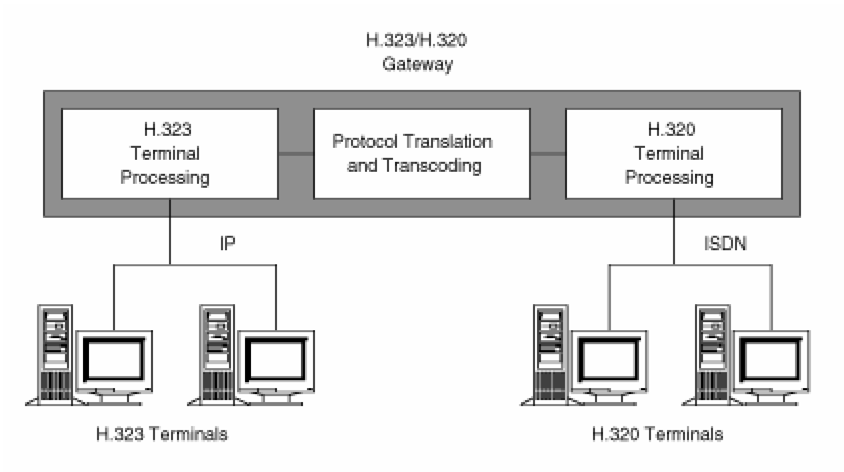
The gatekeeper is an optional component of H.323 that is responsible for managing other components of a H.323 network. It is a very important component to the managed network. The gatekeeper has several responsibilities which include: translation of E.164 aliases to IP addresses, bandwidth management of incoming or outgoing calls, call admission to accept or deny calls, and zone management. Gatekeepers can also support an optional feature that allows a call to be rerouted if there is no answer from the intended terminal. Gatekeepers also help manage different H.323 zones and help manage H.323 MCU sessions. It is important to remember that while gatekeepers are optional, the H.323 terminal must make use of the gatekeeper's services if the gatekeeper is present in the network.

Gatekeepers are typically software products that reside on a server. Although many H.323 MCUs and gateways have embedded gatekeepers, they usually offer less features than stand alone gatekeepers.

There are several gatekeepers that are readily available on the market including the Cisco MCM and the RADVision ECS.

2.1.3 Gateways

If there is a need for a H.323 terminal to communicate with another terminal using H.320, H.324 or analog PSTN networks, a H.323 gateway is required to perform the translation. This optional component typically has ISDN and IP network connections and support the translation between these two networks. The number of simultaneous connections allowed through a gateway is not specified in any standard, so there are different options available from different manufacturers. Gateways may have built-in gatekeepers with minimal features, although this is optional.



There are a several gateway manufacturers with products on the market today. Some good examples are the Cisco 3540 gateway and the RADvision ViaIP gateway.

2.1.4 Multipoint Control Units

The last of the major components is the MCU which controls conferences between 3 or more terminals. The H.323 MCU may be a separate component or may be incorporated into a terminal. The TANDBERG 800, 880, 2500, 6000, 7000 and 8000 have available software options that enable internal H.323 MCU capabilities. The TANDBERG endpoint products are capable of providing MCU services for up to 4 sites. If more capabilities than 4 sites are required, TANDBERG recommends and can provide larger MCUs.

There are a several MCU manufacturers with products on the market today. Some good examples include TANDBERG, Cisco and RADVision.

TANDBERG MCU (front)

TANDBERG MCU (rear)

2.2 Standards

H.323 has its own collection of standards that are defined in the chart below. Other popular communications standards have been included for comparison purposes.

	H.320	**H.321**	**H.322**	**H.323 V1/V2**	**H.324**
Approval Date	1990	1995	1995	1996/1998	1996
Network	Narrowband switched digital ISDN	Broadband ISDN ATM LAN	Guaranteed bandwidth packet switched networks	Non-guaranteed bandwidth packet switched networks, (Ethernet)	PSTN or POTS, the analog phone system
Video	H.261 H.263	H.261 H.263	H.261 H.263	H.261 H.263	H.261 H.263
Audio	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728 G.723 G.729	G.723
Multiplexing	H.221	H.221	H.221	H.225.0	H.223
Control	H.230 H.242	H.242	H.242 H.230	H.245	H.245
Multipoint	H.231 H.243	H.231 H.243	H.231 H.243	H.323	
Data	T.120	T.120	T.120	T.120	T.120
Comm. Interface	I.400	AAL I.363 AJM I.361 PHY I.400	I.400& TCP/IP	TCP/IP	V.34 Modem
Text Chat	T.140	T.140	T.140	T.140	T.140
Encryption	H.233 H.234	H.233 H.234		H.235	H.233 H.234

2.2.1 The Audio Standards:

G.711: 64 Kbps, 8K samples/sec, 8-bit companded PCM (A-law or μ -law), high quality,low complexity. Required for H.320 and H.323.
G.722: ADPCM audio encode/decode (64 kbit/s, 7 kHz).
G.723: Speech coder at 6.3 and 5.3 Kbps data rate. Medium complexity. Required for H.324; Optional for H.323.
G.728: 16 Kbps, LD-CELP, high quality speech coder, very high complexity. Optional for H.320 and H.323.
G.729: 8Kbps, LD-CELP, high quality speech coder, medium complexity. G.DSVD is an interoperable subset.
GSM: Group Special Mobile -- European telephony standard, not ITU. Used by ProShare Video Conferencing software versions 1.0-1.8. 13Kbps, medium quality for voice only, low complexity.

2.2.2 The Video Standards:

H.261: Supports 352x288 (CIF or FCIF) and 176x144 (QCIF). DCT-based algorithm tuned for 2B to 6B ISDN communication. Required for H.320, H.323, and H.324.
H.263: Much-improved derivative of H.261, tuned for POTS data rates. Mostly aimed at QCIF and Sub-QCIF (128x96 -- SQCIF), while providing better video than H.261 on QCIF and CIF. Optional for H.323.

2.2.3 The Communications Standards:

H.221: Frame Structure 64-1920 Kbps.
H.223: Multiplexing protocol for low-bit rate multimedia communication.
H.225: Media Stream Packetization and synchronization on non-guaranteed quality-of service LANs.
H.230: Frame synchronous control and indication signals for audio visual systems.
H.242: System for establishing audio visual terminals using digital channels up to 2Mbps.
H.243: Procedures for establishing communication between three or more audio visual terminals using digital channels up to 2 Mbps.
H.245: Control of communications between visual telephone systems and terminal equipment on non-guaranteed bandwidth LANs.

2.2.4 The Encryption Standards:

H.233: Confidentiality system for audiovisual services
H.234: Encryption key management and authentication system for audiovisual services (Diffe-Hellman key exchange)
H.235: Security and Encryption for H.323 multimedia terminals

3. TANDBERG's IP MCU Implementation

TANDBERG D2 software now enables IP conferencing on the TANDBERG MCU.

Version	Release Date
D2	January 2003

Please consult the appropriate section for details on a particular version of software.

3.1 Common Implementation Features

3.1.1 Remote Management

TANDBERG systems have four methods of remote management.

1. **HTTP**: The MCU has an embedded web server for remote setup and control
2. **HTTPS**: The MCU has an embedded secure web server for secure remote setup and control.
3. **Telnet**: The MCU has an embedded telnet server for full access to the codec's dataport functionality
4. **FTP**: The MCU has an embedded FTP server for uploading software, directories and codec configuration files. In addition, the directory and codec configuration files can be downloaded using FTP.
5. **SNMP**: The MCU has an embedded SNMP manager for proactive reporting of problems and systems status. The TANDBERG Management Suite (TMS) makes use of this feature.

With D2 software, the above services can be disabled.

3.2 D2 Software

3.2.1 Main Features

The main IP features in the D2 release are:

1. H.323 multipoint calls
2. H.323/H.320 mixed multipoint calls
3. Multiple IP conferences
4. H.235 Strong (AES) 128bit encryption
5. H.323 Duo Video
6. H.261+ Custom Formats (XGA, SVGA, VGA, SIF)
7. Multiple video rate transcoding

3.2.2 Layer 4 Ports Used

Function	Port	Type	Direction
Gatekeeper RAS	1719	UDP	⇔
Gatekeeper Discovery	224.0.1.41:1718	UDP	⇔
Q.931 Call Setup	1720	TCP*	⇔
H.245/Q.931	Range 5555—5587	TCP**	⇔
Video	Range 2326—2837	UDP	⇔
Audio	Range 2326—2837	UDP	⇔
FTP	21	TCP*	⇔
Telnet	23	TCP*	⇔
HTTP	80	TCP*	⇔
HTTPS	443	TCP	⇔
NTP	123	UDP*	⇐ (incoming to codec)
SNMP (Queries)	161	UDP*	⇔
SNMP (Traps)	962	UDP	⇒ (outgoing from codec)
Netlog	963	TCP	⇔
FTP/data	1026	TCP	⇔

* denotes a listening port.

** If a site is disconnected and reconnected without terminating the entire conference, the next site to be connected will have an incremented H.245 port. If this functionality is required to function through a firewall, the range of TCP ports can be extended past 5587. However, if a site is disconnected and reconnected a sufficient number of times, without ending the conference, it is possible to quickly exceed this range again.

Outgoing call: *First call uses port 5555 for outgoing Q.931 and port 5556 for H.245, next uses port 5557 for Q.931 and port 5558 for H.245, etc.*

Incoming call: *First uses port 5555 for H.245, second port 5556 etc. Disconnecting a site in a call will not free up available 55XX ports until the whole conference is down.*

3.2.3 Example of and IP call (IP direct dialing)

Using two sites known as 'Site Alice' and 'Site Bob'. Alice will call Bob:

Call Connect:

1. Alice creates a TCP socket for Q.931 and connects to Bob:1720. ***The source port on Alice is 5555.***
2. Bob creates a TCP socket for H.245 and binds it to port 5555
3. Bob sends Q.931 'connect' to Alice and in this message it tells Alice what port the H.245 socket is on
4. Alice creates a TCP socket for H.245 and connects it to the specified port on Bob. ***The source port on Alice is dynamic and cannot be predicted (range 5556 – 5560, depending on the number of sites to be connected)***
5. Capability sets are exchanged
6. Master/slave determination
7. Call is connected

Video Media Stream:

8. Alice creates two UDP sockets, one for RTP and one for RTCP. It binds the RTP socket to an even port number and the RTCP socket to the following odd port number. ***The RTP and RTCP ports cannot be predicted (range 2326—2365)***
9. Alice sends OpenLogicalChannel to Bob which contains the address and port for the RTCP socket
10. Bob creates two UDP sockets, one for RTP and one for RTCP. It binds the RTP socket to an even port number and the RTCP socket to the following odd port number. ***The RTP and RTCP ports cannot be predicted (range 2326—2365)***
11. Bob responds with OpenLogicalChannelAck which contains the addresses and ports for the RTP/RTCP sockets.

Audio Media Stream:

12. Alice creates two UDP sockets, one for RTP and one for RTCP. It binds the RTP socket to an even port number and the RTCP socket to the following odd port number. ***The RTP and RTCP ports cannot be predicted (range 2326—2365)***
13. Alice sends OpenLogicalChannel to Bob which contains the address and port for the RTCP socket
14. Bob creates two UDP sockets, one for RTP and one for RTCP. It binds the RTP socket to an even port number and the RTCP socket to the following odd port number. ***The RTP and RTCP ports cannot be predicted (range 2326—2365)***
15. Bob responds with OpenLogicalChannelAck which contains the addresses and ports for the RTP/RTCP sockets.

Data/FECC Media Stream:

16. Alice creates two UDP sockets, one for RTP and one for RTCP. It binds the RTP socket to an even port number and the RTCP socket to the following odd port number. ***The RTP and RTCP ports cannot be predicted (range 2326—2365)***
17. Alice sends OpenLogicalChannel to Bob which contains the address and port for the RTCP socket
18. Bob creates two UDP sockets, one for RTP and one for RTCP. It binds the RTP socket to an even port number and the RTCP socket to the following odd port number. ***The RTP and RTCP ports cannot be predicted (range 2326—2326)***
19. Bob responds with OpenLogicalChannelAck which contains the addresses and ports for the RTP/RTCP sockets.

3.2.4 Audio

The D2 audio codec of the TANDBERG MCU supports G.711, G.722 and G.728. TANDBERG uses 20ms packets to transmit audio.

Audio	Audio size	IP Header	UDP Header	RTP Header	Total
G.711	160 bytes	20 bytes	8 bytes	12 bytes	200 bytes
G.722	160 bytes	20 bytes	8 bytes	12 bytes	200 bytes
G.728	40 bytes	20 bytes	8 bytes	12 bytes	80 bytes

3.2.5 Video

The D2 video codec of the TANDBERG MCU supports H.261 (QCIF, CIF), H.263 (CIF, 4CIF) and H.263+ (custom formats XGA, SVGA, VGA, and SIF). TANDBERG video is variable and will only take the required bandwidth to transmit the video at any given instant.

Video	Video size (max)	IP Header	UDP Header	RTP Header	Total (max)
H.261	1400 bytes	20 bytes	8 bytes	12 bytes	1440 bytes
H.263/+/++	1400 bytes	20 bytes	8 bytes	12 bytes	1440 bytes

The MTU size can be changed using the 'H323mtu' command in the dataport.

3.2.6 Jitter And Latency

Latency can be defined as the time between a node sending a message and receipt of that message by another node. The TANDBERG MCU can handle any value of latency, however, the higher the latency, the longer the delay in video and audio. This may lead to conferences with undesirable delays causing participants to interrupt and speak over each other.

Jitter can be defined as the difference in latency. Where constant latency simply produces delays in audio and video, jitter can have a more adverse effect. Jitter can cause packets to arrive out of order or at the wrong times. TANDBERG can manage packets with jitter up to 100ms. If excessive packet loss is detected, the TANDBERG MCU will make use of IPLR[TF] and downspeeding to compensate for packet loss

4. TANDBERG MCU H.323 Features

TANDBERG has implemented H.323 version 4 across its entire product line beginning with Version D2 of the TANDBERG MCU software. With this release, TANDBERG has elevated the MCU to a new level, building on such features as Encryption, H.323 MultiSite[TF], H.323 Duo Video[TF], Diffserv, IP Precedence and Type of Service (TOS), RSVP, IPLR[TF] and Simple Network Management Protocol (SNMP).

4.1 Special Features

4.1.1 H.323 MCU

Available on the TANDBERG MCU
From Software Version D2.x to present

The TANDBERG MCU is based on the same robust and feature rich Multisite[TF] found in the TANDBERG endpoints. The MCU is capable of the following number of sites:

	Non-encrypted	Encrypted
128kbps	16	14
256kbps	16	13
384kbps	16	11
512kbps	15	10
768kbps	10	7
2Mbps	4	N/A

The MCU also supports Duo Video[TF], Digital Clarity[TF], QoS, Secure Conference[TF], Downspeeding[TF], IPLR[TF], Polycom's People+Content[TM] and provides new concepts such as Best Impression[TF].

There is support for 3 independent conferences per physical unit.

4.1.2 Intelligent Packet Loss Recovery (IPLR[TF])

Available on the TANDBERG MCU
From Software Version D2.x to present

TANDBERG MCU has a standards based method for improving video quality under packet loss situations. This method works with other vendor's endpoints as well as MCUs. Video quality is improved on the transmit and receive video streams. Please see TANDBERG document D50165 for details on the IPLR feature.

4.1.3 Best Impression^{TF}

TANDBERG's Best Impression^{TF} feature consists of 3 parts:

- Automatic continuous presence(CP) layout switching
- Automatic CP resolution switching
- Enhanced video transcoding.

Automatic Continuous Presence Layout Switching

The TANDBERG MCU has the ability to select several viewing modes with choices of voice switching, CP4, CP9, and CP16. In addition, the TANDBERG MCU offers an 'Auto' mode that will automatically switch between these modes as more sites are added to the conference.

1-2 sites **3-4 sites** **5-9 sites** **10-16 sites**

Automatic Continuous Presence Resolution Switching

Today's MCUs send the entire continuous presence image in CIF (352x288) resolution. This is roughly equivalent to VCR resolution, where each site may have $1/9^{th}$ or $1/16^{th}$ of this already low resolution dedicated to them. Recognizing that 9 or 16 sites on the same screen can lead to poor resolution per site, the TANDBERG MCU will transmit 4CIF (704x576) to all sites when CP9 or CP16 is being displayed. This will give each site 4 times the resolution available on other MCUs currently available on the market.

CIF **CIF** **4CIF** **4CIF**

Enhanced Video Transcoding

The TANDBERG MCU will allow for any video rate, video algorithm, and audio algorithm to be included within the same conference. Further, sites at different rates will be able to see the video quality they expect to see at their rate.

4.1.4 Secure Conference^{TF}

The TANDBERG MCU supports TANDBERG Secure Conference^{TF}. This feature consists of support for both 56bit DES and 128bit AES encryption on both IP and ISDN. AES is currently the highest standards based encryption available to the public. For more information on AES and DES please visit the National Institute of Standards and Technology at www.nist.gov.

4.2 Presentations

4.2.1 H.323 Duo Video^{TF}

Available on the TANDBERG MCU
From Software Version D2.x to present

Duo Video^{TF} is a feature that allows for two simultaneous video streams to be transmitted from one system and to be received by another. This allows a presenter and their content to be seen at the same time, in much the same way you would expect them to be in a local room environment.

If all systems in a conference are TANDBERG dual monitor systems, they will receive the presenter on one screen and the content (PC, Doc Cam, VCR, DVD etc) on the other. If the TANDBERG system is a single monitor system, the video stream will automatically switch from the presenter to the content. This change will take place on the fly with the displayed stream returning to that of the presenter when Duo Video^{TF} is disconnected.

Other manufacturer's systems can be in the same conference when Duo Video^{TF} is used, but they will not see the second stream. With D2, TANDBERG has added support for Polycom's People+Content^{TM} such that if Duo Video^{TF} is initiated by a TANDBERG endpoint, any People+Content^{TM} capable endpoints will receive People+Content. The TANDBERG MCU does not, at this time, support the transmission of People+Content^{TM} from a Polycom endpoint.

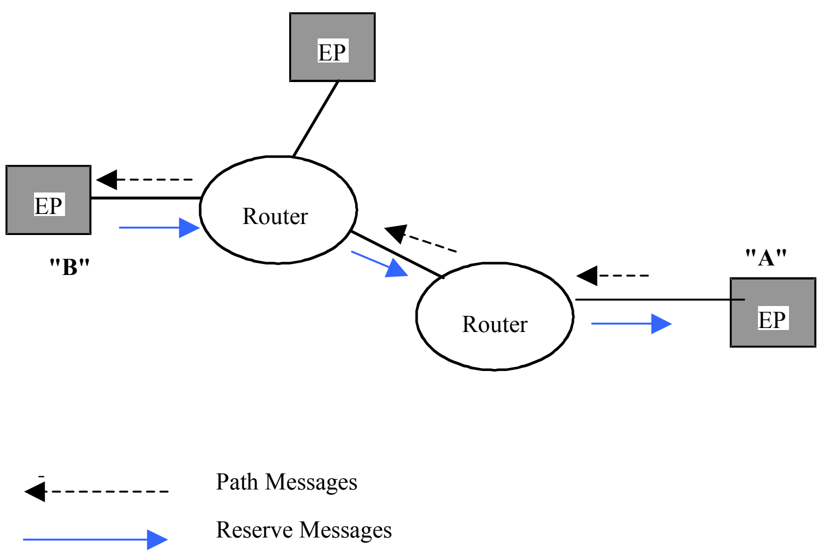
4.3 Quality of Service

4.3.1 RSVP

Available on the TANDBERG MCU
From Software Version D2.x to present

RSVP is specified in the IETF RFC 2205. The RSVP standard is used by an endpoint to request certain qualities from the network that will transport the video and audio data. This request is made at each node throughout the network and each node must comply or the reservation will fail.

The Figure below shows a possible scenario for an RSVP reservation. Endpoint A sends a reservation request (Path message). This request is processed by the routers and forwarded if the reservation is granted. If all routers along the path and endpoint B grants the reservation a Reserve message is sent back to endpoint A.



RSVP is a protocol like TCP and UDP, it has no concept of ports and hence there must be some sort of mechanism in the firewall to enable RSVP traffic. To enable RSVP within a router you need to find the RSVP setting and turn it on. It is advisable to consult your firewall's technical manual to see if there are separate settings for incoming and outgoing RSVP traffic

4.3.2 Differential Services

Available on the TANDBERG MCU
From Software Version D2.x to present

Differential Services is another method of QoS offered by TANDBERG that utilizes 6 bits of the Type of Services Byte. This method is currently replacing IP Precedence as the preferred method for setting priority of packet traffic.

```
  0   1   2   3   4   5   6   7
+---+---+---+---+---+---+---+---+
|         DSCP          |  CU   |
+---+---+---+---+---+---+---+---+

DSCP: differentiated services codepoint
CU:   currently unused
```

4.3.3 IP Precedence and Type of Service

Available on the TANDBERG MCU
From Software Version D2.x to present

TANDBERG has implemented IP Precedence and Type of Service (TOS). IP Precedence allows the video terminal to prioritize its video and audio above or below other IP traffic on the same network

```
Bits 0-2:  Precedence.
Bit    3:  0 = Normal Delay,      1 = Low Delay.
Bits   4:  0 = Normal Throughput, 1 = High Throughput.
Bits   5:  0 = Normal Relibility, 1 = High Relibility.
Bit  6-7:  Reserved for Future Use.

  0     1     2     3     4     5     6     7
+-----+-----+-----+-----+-----+-----+-----+-----+
|   PRECEDENCE    |  D  |  T  |  R  |  0  |  0  |
+-----+-----+-----+-----+-----+-----+-----+-----+

Precedence

111 - Network Control
110 - Internetwork Control
101 - CRITIC/ECP
100 - Flash Override
011 - Flash
010 - Immediate
001 - Priority
000 - Routine
```

The values for signaling, video, audio and data can be independently set in the MCU.

The TOS values available are: Maximum throughput; Minimum monetary cost; Minimum delay; Maximum reliability; Normal (off). This feature is used to allow routers to make decisions concering how best to delay traffic in the event of congestion.

The QoS capabilities of the TANDBERG MCU can be used to manage a private LAN or WAN more effectively. IP Precedence and TOS help prioritize IP traffic, control congestion, and allow for better integration of video conferencing into an existing LAN. During times of high network stress, these features will help the routers to drop lower precedence data to increase router stability and offer higher probability that video conferencing data will reach its destination. In some instances it may help to simply provide a larger proportion of the link without having to configure prioritization policy. In the near future Differential Services may be available from ISP's and ASP's that will use the TOS information to offer better services to the users.

4.4 Remote Management

4.4.1 Simple Network Management Protocol (SNMP)

Available on the TANDBERG MCU
From Software Version D2.x to present

Simple Network Management Protocol (SNMP) is a feature available on most IP based technologies and required by IT professionals to aid in effectively managing their networks. SNMP was developed to reduce the complexities of managing IP based technologies and minimize the amount of resources required to support them. Management programs such as the TANDBERG Management Suite employ SNMP to provide proactive management of video networks. Currently, TANDBERG supports MIB II (RFC 1213) and its own Enterprise MIB expanded upon under MIB II. Details on MIB II can be found at http://www.faqs.org/rfcs/rfc1213.html .

For security reasons, the SNMP manager can be disabled by using the 'services' API command available through the TANDBERG dataport. The SNMP manager can also be password protected using the TANDBERG API command 'ippassword'. Disabling the SNMP manager requires D2 software and later.

4.4.2 HTTP/HTTPS Server

Available on the TANDBERG MCU
From Software Version D2.x to present

The TANDBERG MCU has an integrated web server to allow for remote management of the MCU. To connect to the web server, simply open an Internet browser and type in the IP address of the system to be managed.

For security reasons, the web server can be disabled by using the 'services' API command available through the dataport. The webserver can also be password protected using the API command 'ippassword'. Disabling the HTTP server requires D2 software and later.

4.4.3 FTP Server

Available on the TANDBERG MCU
From Software Version D2.x to present

The TANDBERG MCU has an integrated FTP server to allow for software updates and configuration changes to the MCU. To connect to the FTP server, simply open a FTP client and type in the IP address of the system to be managed.

For security reasons, the FTP server can be disabled by using the 'services' API command available through the dataport. The FTP server can also be password protected using the API command 'ippassword'. Disabling the FTP server requires D2 software and later.

4.4.4 Telnet Server

Available on the TANDBERG MCU
From Software Version D2.x to present

The TANDBERG MCU has an integrated telnet server to allow for system control and configuration changes to the MCU. To connect to the telnet server, simply open a telnet client and type in the IP address of the system to be managed.

For security reasons, the telnet server can be disabled by using the 'services' API command available through the dataport. The telnet server can also be password protected using the API command 'ippassword'. Disabling the telnet server requires D2 software and later.

4.5 Video

4.5.1 H.263+ Custom Video Formats (Digital Clarity^{TF})

Available on the TANDBERG MCU
From Software Version D2.x to present

TANDBERG supports encoding of a true XGA, SVGA, VGA and NTSC (SIF) resolutions. This allows the TANDBERG to send all video signals in their true 'native resolution'. If the source is sending XGA, SVGA or VGA, the receiving system will decode and display the same quality image as was sent by the transmitting side.

Also supported under the H.263+ custom formats is TANDBERG's use of SIF (Source Input Format) that allows for an improved NTSC image over traditional CIF based MCU.

4.5.2 4CIF Digital Clarity[TF]

Available on the TANDBERG MCU
From Software Version D2.x to present

This feature allows for H.263 4CIF (704x576) live video images to be sent to the far end allowing 4 times higher resolution than traditional videoconferencing systems for displaying document camera video and other high resolution images.

4.6 Other

4.6.1 TCS-4

Available on the TANDBERG MCU
From Software Version D2.x to present

TCS-4 allows the user to specify the H.323 alias, in the dial string, of the party being called when calling inbound through a gatekeeper. Currently, if TCS-4 is not supported, the user must use the IVR method by sending DTMF tones after connection to the gateway to specify the E.164 alias of the site to be reached on the LAN. With TANDBERG's implementation of TCS-4, the user can now specify the E.164 alias at the time of the initial call allowing a seamless connection between an H.320 site and a H.323 site.

Example: TANDBERG MCU E.164 alias is 56774
H.323/H.320 gateway ISDN number is 703-555-1212

To dial from an ISDN based TANDBERG endpoint directly to the TANDBERG MCU, simply type 7035551212*56774 into the dial line, then press *Connect* and the call will automatically connect without having to use IVR.

4.6.2 Asymmetrical Encoders and Decoders

Available on the TANDBERG MCU
From Software Version D2.x to present

Realizing there are many different types of videoconferencing units in the world that do not support the same video and audio algorithms, TANDBERG has implemented asymmetrical encoders and decoders. This feature allows different systems with different video and audio algorithms to communicate with the TANDBERG MCU without having to settle on the lowest common denominator. The asymmetrical nature of the MCU really shines when using the mixed mode H.320/H.323 as the TANDBERG MCU can accept *any* of its supported algorithms in any combination from any site and will always transmit the highest possible quality video and audio to the far end.

4.6.3 Inband Changing of Video and Audio Algorithms

Available on the TANDBERG MCU
From Software Version D2.x to present

TANDBERG's MCU is able to change its video and audio algorithms 'on the fly' during a conference. This is valuable when switching between the live user image (CIF) and the live PC (XGA, SVGA, VGA) or Document Camera (4CIF) image. This feature ensures the highest quality picture for each source is displayed.

4.6.4 Flow Control (Downspeeding[TF])

Available on the TANDBERG MCU
From Software Version D2.x to present

The TANDBERG MCU uses Flow Control to control the amount of information being received from each endpoint. If the TANDBERG MCU requires bandwidth to be changed, the TANDBERG MCU will use Flow Control to request the other endpoint to drop the bandwidth.

This feature comes in handy when using H.323 over networks with poor QoS such as the Internet. If the TANDBERG MCU detects excessive packet loss, it will use Flow Control to 'downspeed' the far end overcoming the packet loss problem. Packet loss can occur when routers become overloaded and discard packets or when the receiving video system cannot keep up with the transmitting video system.

4.6.5 Dynamic Bandwidth Management

Available on the TANDBERG MCU
From Software Version D2.x to present

In addition, TANDBERG understands how valuable the IP network is to the user and how valuable the bandwidth is to the processes that take place on that network. TANDBERG will dynamically change the bandwidth used based on bandwidth required during a call. If the full bandwidth is not required, i.e. low motion video, the bandwidth used by the system will be decreased, freeing up this bandwidth for other processes such as Internet, email, VoIP and other applications currently sharing the corporate network.

4.6.6 Lip Sync

Available on the TANDBERG MCU
From Software Version D2.x to present

Synchronizing audio to video across synchronous networks such as ISDN is a task most video conferencing units today are capable of doing very well. However, IP is an asynchronous network that sends audio and video separately. It is easy to see that these two streams may not arrive at their destination at the same time. Lip sync problems are a certainty if precautions are not taken in the implementation of the codec.

TANDBERG supports sequencing of the video and audio IP packets and the reassembly and reordering of these packets at the destination if they are received out of order. If the video stream is received before the audio stream, the TANDBERG MCU will buffer this data until the necessary audio is received and reassemble the data that the destination codec will use to reproduce a clear image with exceptional sound quality and lip sync.

5. Using H.323 With TANDBERG

It is important to understand exactly how H.323 communications take place. Without this understanding, it is next to impossible to create a solution that will be flexible enough to compete with ISDN technology in terms of ease of use. To begin, we must examine the elements of session establishment, handshaking and exchange of audio, video and data.

It is also important to remember a gatekeeper is strongly recommended if a gateway or MCU is a component in your network design.

There are 3 types of gatekeepers available:

- Direct signalling and direct media
- Routed signalling and direct media
- Routed signalling and routed media.

The most common gatekeeper in use today is the Routed signalling and Direct media gatekeeper. The next few examples describe calls using Routed signalling and Direct media gatekeepers.

5.1 Only Calling Endpoint Has Gatekeeper

Endpoint 1 (EP1) is registered to a local Gatekeeper 1 (GK1). Endpoint 2 (EP2) is not registered to a gatekeeper. Gatekeeper is using routed signalling and direct media.

Step	Function	Translation
1	EP1 issues Admissions Request (ARQ) to GK1 using RAS	EP1 asks permission from GK1 to call EP2
2	GK1 issues Admission Confirm (ACF) to EP1 using RAS	GK1 gives EP1 permission to call EP2
3	EP1 sends Q.931 'setup' message to GK1	EP1 dials the IP address of EP2 and sends it to GK1
4	GK1 sends Q.931 'setup' message to EP2	GK1 dials the IP address of EP2
5	GK1 sends Q.931 'call proceeding' message to EP1, EP2 sends Q.931 'call proceeding' message to GK1	GK1 tells EP1 the call is in process. EP2 tells GK1 the call is in process
6	EP2 sends Q.931 'alerting' message to GK1, GK1 sends Q.931 'alerting' message to EP1	EP2 tells the GK1 it is ringing. GK1 tells EP1 that EP2 is ringing
7	EP2 sends Q.931 'connect' message to GK1	EP2 tells GK1 that is has connected.
8	GK1 sends Q.931 'connect' message to EP1	GK1 tells EP1 that EP2 has connected.

Video, Audio, and FECC media is passed directly between EP1 and EP2.

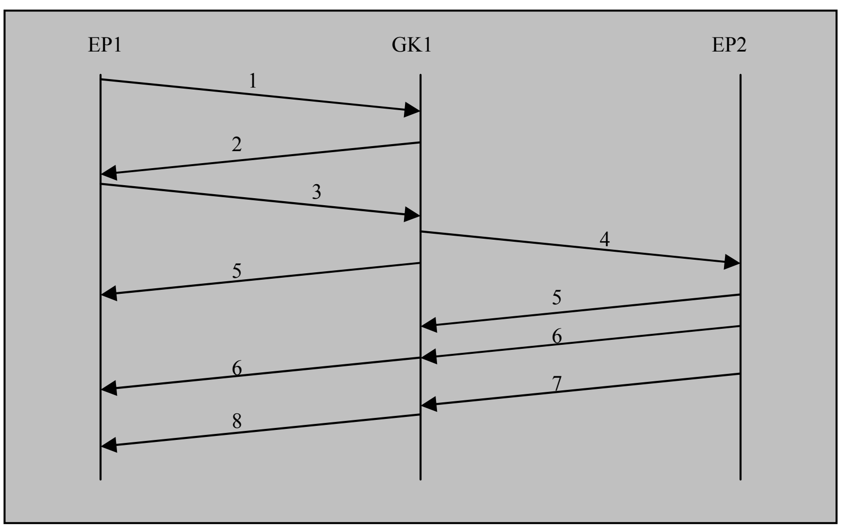


FIGURE 1

5.2 Only Called Endpoint Has Gatekeeper

Endpoint 1 is not registered to a gatekeeper. Endpoint 2 is registered to GK2 and is using routed signalling and direct media.

Step	Function	Translation
1	EP1 sends Q.931 'setup' message to EP2	EP1 dials EP2
2	EP2 sends Q.931 'call proceeding' message to EP1	EP2 tells EP1 the call is proceeding
3	EP2 sends Admissions Request (ARQ) to GK2 using RAS	EP2 asks permission to accept the call from GK2
4	GK2 sends Admissions Confirm (ACF) to EP2	GK2 denies call and requests to route call through gatekeeper
5	EP2 responds to EP1 with Call Signalling Transport Address of GK2	EP2 tells EP1 where to send the call signalling info
6	EP1 issues Q.931 'release complete' message to EP2	EP2 hangs up the current call
7	EP1 sends Q.931 'setup' message to GK2	EP1 dials GK2 Call Signalling Transport Address
8	GK2 sends Q.931 'setup' message to EP2	GK2 dials EP2
9	EP2 issues Admissions Request (ARQ) to GK2	EP2 asks for permission to answer call from GK2
10	GK2 issues Admissions Confirm (ACF) to EP2	GK2 gives permission to answer call to EP2
11	EP2 sends Q.931 'alerting' message to GK2, GK2 sends Q.931 'alerting' message to EP1	EP2 tells GK2/EP1 it is ringing
12	EP2 sends Q.931 'connect' message to GK2	EP2 tells GK2 it is connected
13	GK2 sends Q.931 'connect' message to EP1	GK2 tells EP1 it is connected

Video, Audio, and FECC media is passed directly between EP1 and EP2.

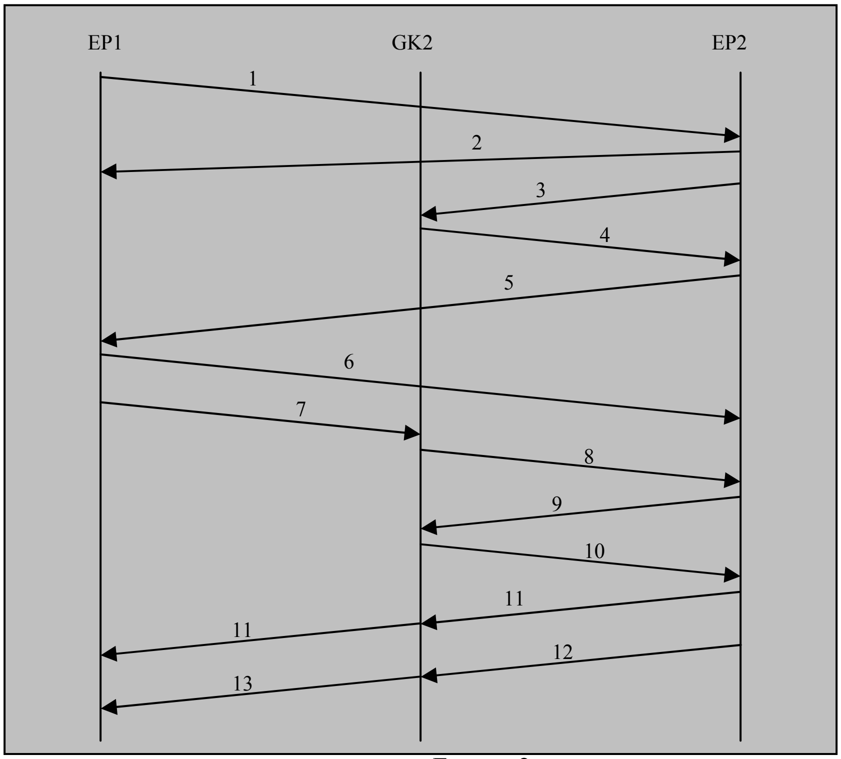


FIGURE 2

5.3 Both Endpoints Registered To Different Gatekeepers

Endpoint 1 is registered to GK1 and Endpoint 2 is registered to GK2. Both are using routed signalling and direct media.

Step	Function	Translation
1	EP1 issues Admissions Request (ARQ) to GK1	EP1 asks permission to call from GK1
2	GK1 issues Admissions Confirm (ACF) to EP1	GK1 gives permission to call to EP1
3	EP1 sends Q.931 'setup' message to GK1	EP1 dials EP2 through GK1
4	GK1 sends Q.931 'setup' message to EP2	GK1 dials EP2
5	EP2 sends Q.931 'call proceeding' to GK1, GK1 sends Q.931 'call proceeding' to EP1	EP2 notifies GK1/EP1 that the call is proceeding
6	EP2 issues Admissions Request (ARQ) to GK2	EP2 asks for permission to answer call from GK2
7	GK2 issues Admissions Confirm (ACF) to EP2	GK2 will deny and ask to route call to gatekeeper
8	EP2 sends GK1 Call Signalling Transport Address of GK2	EP2 tells GK1 where to call GK2
9	GK1 issues Q.931 'release complete' to EP2	GK1 hangs up the current call
10	GK1 issues Q.931 'setup' message to GK2	GK1 calls GK2
11	GK2 issues Q.931 'setup' message to EP2	GK2 calls EP2
12	EP2 issues Admissions Request (ARW) to GK2	EP2 asks for permission to answer call from GK2
13	GK2 issues Admissions Confirm (ACF) to EP2	GK2 gives permission to answer call to EP2
14	EP2 sends Q.931 'alerting' message to GK2, GK2 sends Q.931 'alerting' message to GK1, GK1 sends Q.931 'alerting' to EP1	EP2 tells GK2/GK1/EP1 it is ringing
15	EP2 sends Q.931 'connect' message to GK2	EP2 tells GK2 it is connected
16	GK2 sends Q.931 'connect' message to GK1	GK2 tells GK1 that EP2 is connected
17	GK1 sends Q.931 'connect' message to EP1	GK1 tells EP1 that EP2 is connected

Video, Audio, and FECC media is passed directly between EP1 and EP2.

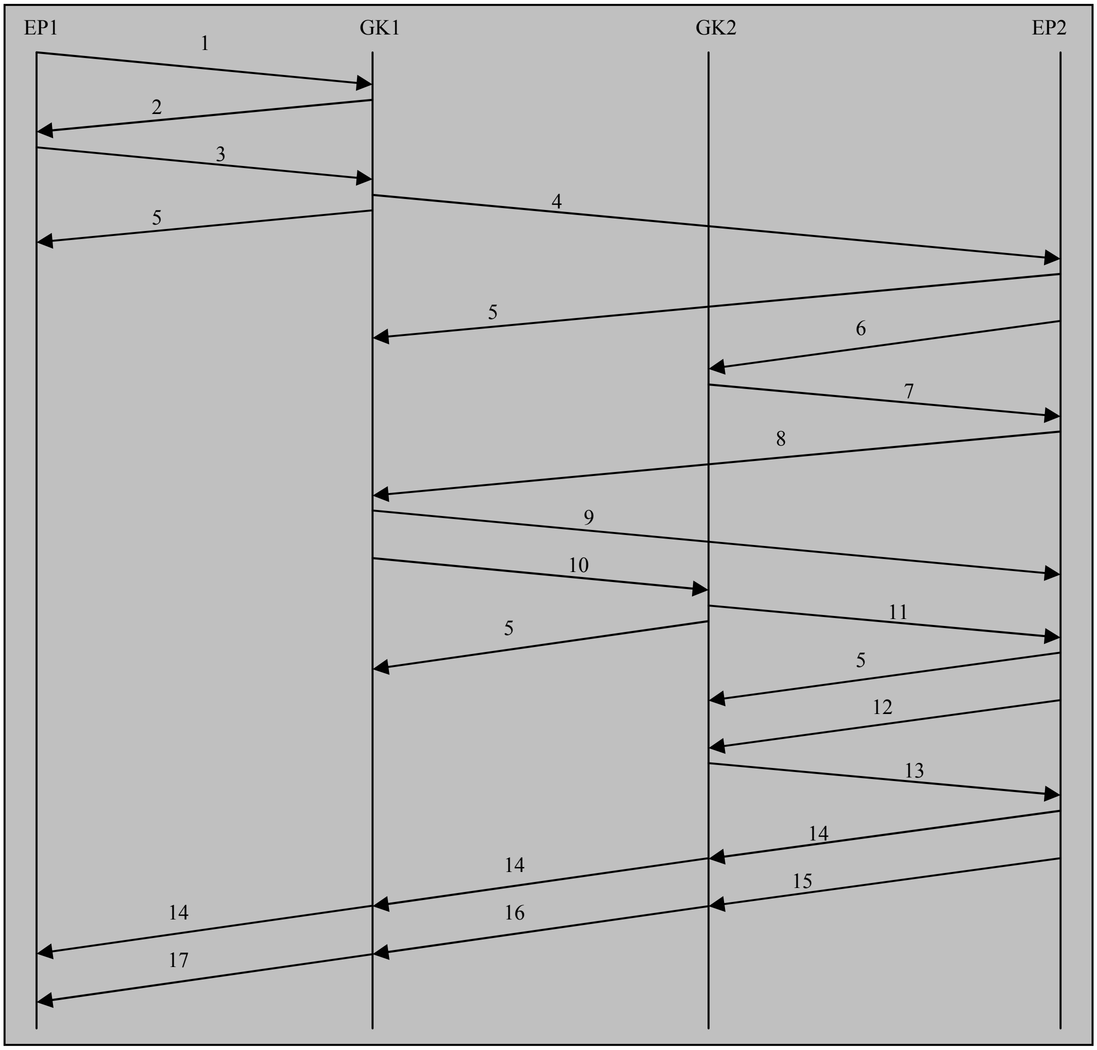


FIGURE 3

6. List of Terms

ASP – Application Service Provider

Ethernet – A local-area network (LAN) protocol developed by Xerox Corporation, DEC and Intel in 1976. Ethernet uses a bus or star topology and supports data transfer rates of 10 or 100 Mbps. It is one of the most widely implemented LAN standards.

H.225 – ITU Standard that describes H.323 call establishment and packetization. This standard also describes the use of RAS, Q.931 and RTP.

H.245 – ITU Standard that describes H.323 syntax and semantics of terminal information messages as well as procedures to use them for in-band negotiation at the start of or during communication.

H.323 – ITU Standard that describes packet based video, audio and data conferencing on networks with non guaranteed Quality of Service (QoS).

ISP – Internet Service Provider

Jitter – Jitter is the variation in network latency. Typically, video systems should be able to accommodate jitter up to at least 100ms.

LAN – Local Area Network.

Latency – The time between a node sending a message and receipt of the message by another node. Typically any latency is supportable, providing it is constant, but large latencies may result in a poor videoconference.

Packet Loss – Occurs when data is lost from the bit-stream, typically on public networks such as the Internet. Packet Loss can occur when passing through a router and has a higher chance of occurring as the hop count is increased. Packet loss can also occur at the receiver end when the transmitter sends data too quick.

Port – In TCP/IP and UDP networks, an endpoint to a logical connection. The port number identifies what type of port it is. For example, port 80 is used for HTTP traffic.

Q.931 – Used to signal call setup on ISDN. Also used by H.225 to establish and disconnect H.323 calls.

RAS – Registration, Admission and Status Protocol. Used by endpoints and gatekeepers to communicate.

RSVP – Resource Reservation Protocol for reserving bandwidth through a RSVP enabled IP network.

RTCP – Real Time Control Protocol. RTCP provides a mechanism for session control and has four main functions: quality feedback, participant identification, RTCP packet transmission rate control and session control information transmission. The primary function of RTCP is to provide feedback

RTP – Real Time Protocol. Described by H.225 on how to handle packetization of audio and video data for H.323. RTP does provide information to reconstruct real time data such as: payload type identification, sequence numbering and timestamping. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services

TCP – Transport Control Protocol. A connection oriented Layer 4 protocol used in H.323 to connect Q.931 and H.245 streams.

UDP – User Datagram Protocol. A connectionless protocol used in transmission of data over IP. While it does not require as much overhead as TCP, it is not as reliable in delivering data. UDP is used to transmit audio and video data in H.323

7. Appendix A

7.1 Polycom H.323 Endpoints Layer 4 ports

This information is provided 'as is' and is not guaranteed to be accurate as it is based on information provided by Polycom. If the version of Polycom software allows for fixed port specification, the following ports should be opened:

Function	Port	Type	Direction
Gatekeeper (RAS)	1719	UDP	⇔
Gatekeeper Discovery	224.0.1.41:1718	UDP	⇔
Q.931 Call Setup	1720	TCP*	⇔
H.245*	Range xxxx—3231**	TCP	⇔
Video	Range xxxx—3235**	UDP	⇔
Audio	Range xxxx—3235**	UDP	⇔
Data (Far End Camera Control)	Range xxxx—3235**	UDP	⇔

** The lower limit of these port ranges are user specified.

If the version of software from Polycom does not support user specified ports then the following ports should be opened:

Function	Port	Type	Direction
Gatekeeper (RAS)	1719	UDP	⇔
Gatekeeper Discovery	224.0.1.41:1718	UDP	⇔
Q.931 Call Setup	1720	TCP*	⇔
H.245*	Range 1024—65535	TCP	⇔
Video	Range 1024—65535	UDP	⇔
Audio	Range 1024—65535	UDP	⇔
Data (Far End Camera Control)	Range 1024—65535	UDP	⇔

7.2 PictureTel H.323 Endpoints Layer 4 ports

This information is provided ‘as is’ and is not guaranteed to be accurate as it is based on information provided by PictureTel.

Function	Port	Type	Direction
Gatekeeper RAS	1719	UDP	⇔
Gatekeeper Discovery	224.0.1.41:1718	UDP	⇔
Q.931 Call Setup	1720	TCP*	⇔
H.245	Range 1700-1750	TCP	⇔
Video	Range 17000-17050	UDP	⇔
Audio	Range 17000-17050	UDP	⇔
Other UDP	Range 1700-1750	UDP	⇔