Wyoming Telehealth Consortium

Video Conferencing
Technical Policies and Procedures

May 2011

Table of Contents

Introduction

The Wyoming legislature appointed a consortium of health stakeholders and organizations to create Telehealth Policy and Procedure for applications in Wyoming. Documents which should accompany this document include the Wyoming Telehealth Provider Guide, the End User Troubleshooting Guide, the Polycom Quick Guide, and more. Information can also be found on the Southeast Wyoming Telehealth Network website at www.sewtn.net

Room Considerations

Both the physician video conferencing room and the patient video conferencing room should adhere to normal exam room requirements based on HIPAA requirements and policies for your clinic or hospital. Treat patient privacy to the same level as if you were seeing the patient in person.

Recommended Room Color: Benjamin Moore paint numbers 1627 or 829 This light blue-grey contrasts well against skin tone.

Microphone placement should be close to the patient to receive appropriate patient intake in an Exam Room. Microphone placement should be central to all participants and the use of an omni-directional flat mic works well in small to medium groups. Permanent fixtures such as ceiling mount microphones should be considered for auditorium-like applications or for very large conference rooms. Ceiling mount microphones should not be mounted near in-ceiling speakers, PA announcement systems, HVAC vents, or ceiling mount fans; these items will create ambient noise which will be distracting for a video conference.

Lighting should be controlled room lighting and not natural lighting. Natural lighting from windows will cast heavy shadows and other uncontrollable effects. Windows should be covered or blinds closed during a video conference. Video conferencing equipment should not be facing windows as this will cause attendees to look like 'silhouettes'. If overhead lighting is not adequate, provide ambient can-lighting fixtures near a wall. These fixtures sit on the floor and point upwards to provide additional ambient lighting and are very inexpensive.

Lighting considerations are especially true when working in an Exam Room environment as patient examination is critical.

Please refer to the Telehealth Provider document for more detailed information.

Video Conferencing Endpoints

System Requirements for H.323 Systems:

The ITU-T H.323 protocol standards should be adopted for Wyoming telehealth video conferencing as it is an industry standard. The following minimum requirements should be included for Room Based video conferencing units as well as H.323 based Desktop video conferencing implementations.

H.323 Stack Minimum Requirements

- IPv4/IPv6 Support

- H.460.17-19 Firewall/NAT Traversal
- UDP Signaling (Annex E)
- HTTP-Based Service Control for H.323 Devices (Annex K)
- Telephony Signaling Tunneling Through H.323 (Annex M)
- DNS support (Annex O)
- Remote Camera Control (Annex Q)
- H.341 MIB Support
- Q.931 Multiplexing
- High Capacity Registration (Additive Registration)
- H.245v13 Advanced Call Control
- H.235v3 Security
- Full H.450 Supplementary Services
- H.245 GEF API
- H.350 LDAP support

System Configurations for H.323 Room Based Systems:

In addition to room considerations and network considerations:

- All telehealth video units should be registered to a public gatekeeper. This gatekeeper will provide a registered alias for direct dialing.
- Video conferencing equipment must be kept up to date by using vendor supplied annual maintenance. Video conferencing equipment that is two (2) revisions of software older or no longer supported by the industry (end of life) should be replaced.
- Video conferencing equipment will be given a static IP address, DHCP is not an option.
- Duplex and Link set to 100 or 1000 Full
- SNMP community is set to: wythealth
- Qos is set for Diffserv: Audio: 46, Video: 34, Data and Signaling: 26
- Each video conferencing endpoint will be set with an administrator password and provided only to pertinent video conferencing *IT administrators*.

System Configuration for H.323 Desktop Video Conferencing

Desktop video sessions use the CMA Desktop application or Polycom PVX application. These applications allow for H.323 video conferences to occur using a web cam, instead of a room based application.

POLICY:

The CMA Desktop contains 300 licenses. In order for outside entities to use a CMA Desktop license, the CRMC video network administrator will need to know the name, address, phone, and email of the individual who would like to use it. The video network administrator will create a username and password for the individual. This will be documented.

PROCEDURE:

- The following is the configuration for a CMA Desktop installation:
  - Registration to: CRMC's external Video Border Proxy External Border Proxy: 65.121.101.125
  - The same firewall ports will be necessary for desktop video conferencing as room based video conferencing systems. Please refer to Appendix A for an overview of common video conferencing ports used.
  - Duplex and Link set to 100 or 1000 Full
  - Qos is set for Diffserv: Audio: 46, Video: 34, Data and Signaling: 26
  - Username and Password is created by the video conferencing administer on the CMA 4000 Server and will be provided to the end user.

- Alias Naming Scheme:

  - CMA Desktop Installation: the phone extension of the user
  - In the event of a duplication, add a 1 or successive number to the end of the extension
  - Those on the CRMCWY Network *and* that have an entry in the Outlook Directory will have an alias of their phone number in Outlook

Video Network

Video Network Consideration

The minimum network requirements for video conferencing at each hospital or clinic:

- Minimum bandwidth dedicated to basic video conferencing: 402k
- This number is the minimum, per video conferencing unit, that should be implemented for basic video conferencing which is defined as "people talking with possible H.239 graphics sharing".
- Minimum bandwidth dedicated to advanced video conferencing: at or above 2 Mbps
- This number is the minimum, per video conferencing unit, that should be implemented for advanced video conferencing which is defined as "high definition video of people talking with possible H.239 graphics sharing and advanced peripherals needed for specialty practice".
- Managed Switch at the video conferencing segments – Layer 3 aware
- Duplex and Link set to 100 or 1000 Full
- Qos is set for Diffserv: Audio: 46, Video: 34, Data and Signaling: 26

Gatekeeper Functionality

For gatekeeper the ITU-T H.460.17, H.460.18, and H.460.19 protocol standards should be adopted for telehealth video conferencing in Wyoming.

- H.225 for RAS and Registration
- Q.931 for setup and connection
- H.245 for negotiated media channel

- H.323 Firewall / NAT Traversal
- H.323 / ALG aware Firewall

Video Conference Calling

POLICY:

- Point to point calls are accepted and encouraged. Point to point calls can be made within the network or with outside entities. Point to point calls within the network may be made by Alias dialing by using the alias, or may be made with the IP address. Video units which contain a Multi-Point feature may place point to point calls with up to 4 entities at one time (license permitting on that specific video unit). However, in most cases if more than 2 video units would like to be involved in a video call the bridge will need to be scheduled to launch the call to all participants.
- Bridging calls are accepted and encouraged. Bridging is ideal when more than 2 participants would like to be involved in a video call.
-
- The bridge is a Polycom MGC-50:
    - Standard Video:
        - 96 ports for 128k and 256k
        - 48 ports for 384k and 512k
        - 24 ports for 768k and 1152k
        - 12 ports for T1
    - Standard Video with Continuous Presence:
        - 12 ports for 384k and 512k
    - Standard Video with Encryption:
        - 48 ports for 128k
        - 36 ports for 256k
        - 24 ports for 384k
        - 18 ports for 512k
        - 12 ports for 768k
        - 9 ports 1152k
        - 6 ports for T1

  - The bridge may also contact video units within the network or with outside entities. The bridge may dial video endpoints using Alias dialing or IP address. Bridge Cascading is not recommended, but may provide a connection method when direct contact with a video endpoint is not possible.

PROCEDURE:

- Scheduled Telehealth Bridged calls are scheduled with the following technology guidelines:
  - Bridge will dial outbound to all locations 15 minutes prior to event start time
  - Call speed: 384k
  - Network Layout: Standard, Voice Switched (unless requested otherwise)
  - FECC (far-end camera control on)
  - H.239 Graphics set to On
  - AES Encryption set to Off
  - Auto Video compression (to accommodate current and legacy video equipment)
  - Auto Audio compression (to accommodate current and legacy video equipment)
  - The organizer of the video event should also specify date, start and end times, lecture mode, continuous presence, or recurrence of event. Speak to the person in charge of scheduling to learn how.
  - Meeting rooms are always available.
  -

Video Conference Call Methods

Have Someone Call You

Other video networks may call you directly, via video. In order to do this the following must occur:
- You must provide the far-site your video number IP address
- The video unit must be plugged in; with power and internet access

How to Call Others

You may call other video sites, there are several different ways to do this.
- You will need to know which method to use to call the far-site (please see examples below).
- The far-site must provide you a good, public, IP address for their video conferencing unit.

Method 1: Direct IP Address

Video conferencing networks use a Public IP address. Each address is unique. You will need the specific number from the person you want to call. You may need to ask their IT person for this information. An IP address is a sequence of numbers separated by four periods, such as: 64.238.3.15

Non-Routable IP Address: If you are given the following IP addresses the call will fail as these IP addresses are not public: 10.x.x.x, 192.168.x.x, 172.x.x.x (x can be any number)

Method 2: Alias Dialing

While all video networks are set up differently, some may choose to use an Alias for dialing. Each address is unique. You will need the specific number from the person you want to call. You may need to ask their IT person for this information.   An alias will look like a string of numbers, then an '@' symbol, then a public IP address. This routes the call to the proper endpoint on another network. An alias address will look like: 242020@65.121.101.125 or 65.121.101.125##242020.

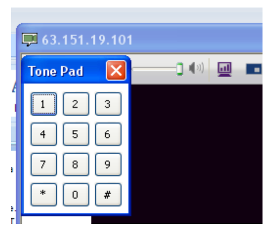
Method 3: Dialing in to a Bridge

In some cases you may be part of a larger conference which uses a bridge. In this case, if you are asked to dial in to their bridge ask which method they would like you to use. Each address is unique. You may need to ask their IT person for this information. They may want you to dial in to their bridge using:

- Direct IP address which looks like 159.238.3.15
- Alias Dialing which looks like 1004@65.121.59.12
- Direct IP address with Room Entry Code which looks like 159.238.3.15##40

If you need to use a Room Entry Code (sometimes referred to as DTMF code) do the following:
Enter the IP address you have been instructed to call (which looks like 159.238.3.15) and choose Call.

Once in the call, choose the Tone Pad from the main menu.
Enter the conference number or entry code you have been given.
Once in the conference close the Tone Pad window.



CMA Desktop Clinical selection considerations

Cost:

For clinical applications, there are several needs that need to be met in order to bring video conferencing successfully into a clinic environment. Among the most important is that the endpoint costs need to be low, as every provider has several exam rooms and his own desktop. CMA Desktop is not a computer resource hog. Almost any computer that is from 2-3 years old to any new computer today will have the capacity to facilitate video conferencing. That being the case, it should be almost no cost to find a computer to load the application onto. The software application is free and a high definition web camera is less than $100. This is a very compelling cost proposal to health care providers at this cost of entry level.

Physical space:

With EMR being mandated across the country, either every provider will have a computer or every exam room will have a desktop to facilitate EMR. These very same computers can have the zero cost CMA Desktop application installed. This effectively places video conferencing in every Dr's office and clinic room for zero cost and for not any additional desk space or room required in the exam room.

Availability:

In the recent past it was thought necessary to have equipment for telehealth video conferencing mounted to a cart that was stored somewhere and wheeled in when needed. The size of the cart, relative to a small exam room and the disruption in clinic flow, were large barriers to use. The $20,000 ea video codec cost was a barrier to having them available at all. Today, to be available for a video conference in any exam room in a clinic, the only additional item needed would be to plug in a web cam.

Image:

With technology advances, today's high definition web cam offers superior images to expensive codecs that are just a few years old. For many years other telehealth networks have been using standard definition cameras to deliver telehealth. Today's availability of low cost, high quality images should help diminish any providers concern over visually missing some important aspect of the exam.

Collateral equipment:

There continues to be additional clinical exam support equipment for telehealth. The great news is that with the PC being centric to the exam, the computing power enables many functions that, in the past with dumb video codec's, required separate boxes. Today we have wireless stethoscopes, USB enabled ECG's, and exam cameras. There have been a few pc based ultrasound devices. This availability of PC centric devices should continue to expand.

Infrastructure:

What enables the endpoint cost to be so low is the infrastructure already in place for classic video conferencing along with the free Polycom CMA desktop application. With Polycom, a CMA 4000 is necessary to manage video bridge conferencing. By adding equipment to enable external access via the CMA 4000 we are able to acquire and then offer low cost per user licenses.

CMA Download and Configuration

Download the 5.1 version of CMA desktop and install it.

Goto:

`http://support.polycom.com/PolycomService/support/us/support/video/cma/cma_desktop.html`

You might need a program to unzip this download. I have used www.7-zip.org successfully, and it is free. During the installation it will ask if you want a flexible IP address or a specific one. Check specific and use this IP 65.121.101.125

Accept anything the installer wants to do. Once everything is installed launch the CMA desktop. Your user name will start with "Local\" without the quotes. Your logon id should look like:

Local\xxxxxx Capital letters are recognized there is a space between first and last name.

State of Wyoming TeleHealth/TeleConference Standards

The use of the term “standards-based” and “consumer-grade” to define the different videoconferencing markets may result in some concerns or confusion. After all, some consumer-grade products use standards for video encoding, while other standards-based systems may not implement all of the possible videoconferencing standards.

Below is a collection of many of the standards that are pertinent to videoconferencing, with the caveat that they focus on IP-based, packet-switched networks and products as opposed to ISDN-based or telephony-based systems. You will find a description of the standards below.

Multimedia Call Control Standards – SIP and H.323

These two standards help initiate, manage, and terminate audio-video communications on networks. SIP and H.323 do not communicate between one another, although interoperability can be established with a gateway device, which helps translate between the two protocols. A standard included within the H.323 standard is H.245, which serves as the primary call control “handshake” that occurs between devices at the start of a videoconferencing session. The H.245 standard has been updated to include a faster call control protocol based on the H.255 standard, supporting something called the “Fast Connect” procedure.

Video Standards – H.263 and H.264

These two standards are used to compress video, specifically in this context to reduce bandwidth when sending video data over a network between two video systems. H.264 provides significant improvements in compression over H.263 and older H.26X standards. H.264 is also referred to as H.264/AVC for “Advanced Video Coding,” or Single-Layer H.264.

Audio Standards – G.711, G.722, G.729

These standards are used for companding audio in video conferencing. G.711 is the standard required by H.323, whereas G.722 and G.729 are optional. The primary difference between these standards is the sampling frequency and compression of the audio. G.722 provides improvements by doubling the sampling frequency of the audio when compared to G.711, which results in a potential improvement in the quality and clarity of the received audio, but an increase in the required bandwidth. G.729 requires less bandwidth by providing a less literally accurate transmission of sound that has been optimized for speech. This may make speech sound clearer, but less true to the actual voice data (and likely to be non-ideal for medical diagnosis, such as electronic stethoscopy).

Testing to confirm teleconferencing capability interoperability

There are two types of dialing supported by video conferencing bridges. End point to end point dialing and meeting room dialing. These tests, to be considered valid for interpretational reasons, need to be performed between two or more differing manufactures equipment. Today the major manufactures are Polycom, Cisco, Radvision, and Logitech.

Open end point to endpoint dialing. Both audio and video connections need to be present and stable.

1. Direct IP dialing. This test should be performed bidirectionally. That is defined for these purposes that one video codec is first an endpoint originator of the call and then for the second part of the test it is the receiving endpoint of the call. This test is for video codecs that typical have static IP address either in the open or behind a firewall. Obviously firewall traversal and NAT may play a pivotal role in this test. The test will consist of entering a ip address and successfully connecting with endpoint.

2. Direct IP dialing with the addition of an alias. This is the same test as above with the added requirement that the aliased endpoint can be reached. The ideal situation would be a single string of numbers and characters that would result in and endpoint being connected. The reason for this is to be able to build global address books that span differing systems but require easy use by a user with no knowledge of which system they are dialing into. With a single click the user would connect two endpoints. The test would consist of entering a single string of characters that includes an IP address and successful connecting with another endpoint. with additional characters or numbers

3. It needs to be determined by the user community that if first connecting to and entry cue and then dialing an alias is acceptable. It is not known if this can be a part of a address book entry.

Meeting room dialing.

Meeting rooms are used sometimes because an unknown number of people attending a conference or the people joining may not be known in advance. Other reasons are to give a persistent electronic place to conduct business or deliver care. There are no tests that need to be applied for meeting rooms that are on the same network. For connecting with a meeting room externally the connection scheme should be the same as above. Either direct IP dial or dial with an alias.

Encrytpion.

Video conferencing equipment is expected to be able to negotiate an encrypted connection using AES 128 bit encryption algorithm between two different systems.

Appendix A

Well Known Port Numbers Used in Videoconferencing

AVI Systems, Inc.

Please note: this listing is a basic list of ports used for H.323 video conferencing. However, some makes, models, and codec software versions may require specific ports not listed here. In addition, registering video conferencing codecs to video conferencing infrastructure may require additional ports or configurations based on network design. Please refer to the latest whitepaper for your specific make and model of video conferencing equipment.

Port	Type	Protocol	Application	Manufacturer
21	Static	TCP	File Transfer Protocol for endpoint software upgrades (must be bi-directional)	Polycom and Tandberg
23	Static	TCP & UDP	Telnet (must be bi-directional)	Polycom, Sony, Tandberg
80	Static	TCP	Hypertext Transfer Protocol (HTTP) – web browser interface for codec control and menus	Polycom, Sony, Tandberg
161	Static	UDP	Simple Network Management Protocol (SNMP) Queries	Tandberg
389	Static	TCP	Lightweight Directory Access Protocol (LDAP) – ILS registration	Polycom
962	Static	UDP	Simple Network Management Protocol (SNMP) Traps	Tandberg
963	Static	TCP	This port is not assigned, but Tandberg uses it for Netlog	Tandberg
964	Static	TCP	This port is not assigned, but Tandberg uses it for FTP/data	Tandberg
965	Static	TCP	This port is not assigned, but Tandberg uses it for VNC	Tandberg
970	Static	UDP	This port is not assigned, but Tandberg uses it for Real-time Transport Protocol (RTP) for streaming video	Tandberg
971	Static	UDP	This port is not assigned, but Tandberg uses it for Real-time Transport Control Protocol (RTCP) for streaming video	Tandberg
972	Static	UDP	This port is not assigned, but Tandberg uses it for Real-time Transport Protocol (RTP) for streaming audio	Tandberg
973	Static	UDP	This port is not assigned, but Tandberg uses it for Real-time Transport Control Protocol (RTCP) for streaming audio	Tandberg
974	Static	UDP	This port is not assigned, but Tandberg uses it for SAP	Tandberg
1002	Static	UDP	This port is not assigned, but Vcon uses it for Lightweight Directory Access Protocol (LDAP) – ILS registration	Vcon

Registered Port Numbers Used in Videoconferencing

Range	Type	Protocol	Application	Manufacturer
1300	Static	TCP & UDP	This port is registered to Intel and is used to secure a H.323 host call – h 323hostcsllsc (must be bi-directional)	Polycom
1503	Static	TCP	This port is registered to Databeam and is used for T.120 file sharing	Polycom, Sony, Tandberg and Vcon
1718	Static	TCP & UDP	This port is registered to Intel and is used to secure a H.323 host call – h 323gatedisc (must be bi-directional)	Polycom, Sony, and Vcon
1719	Static	TCP & UDP	This port is registered to Intel and is used foe gatekeeper RAS – h 323gatestat (must be bi-directional)	Polycom, Sony, Tandberg and Vcon
1720	Static	TCP & UDP	This port is registered to Intel and is used to establish a H.323 host call using Q.931 call setup – h 323hostcall (must be bi-directional)	Polycom, Sony, Tandberg and Vcon
1731	Static	TCP & UDP	Audio call control –msiccp – for VoIP	Polycom
1024 – 65535				Vcon
1024 – 65535				
2253 – 2255	Dynamic		Sony uses an available port in this range for the exchange of H.245 call parameters. (Also known as RTCP)	Sony
2326 – 2373	Dynamic	UDP	Tandberg uses an available port in this range for video data streams	Tandberg
2326 – 2373	Dynamic	UDP	Tandberg uses an available port in this range for audio data streams	Tandberg
2326 – 2373	Dynamic	UDP	Tandberg uses an available port in this range for data transfers and Far End Camera Control – FECC	Tandberg
2979	Static	TCP & UDP	This port is registered to ACM for H.263 Video Streaming	Polycom
3230 – 3247	Dynamic	UDP	Polycom uses an available ports in this range for audio and video	Polycom
3230 – 3235	Dynamic	UDP	Polycom uses an available port in this range for the exchange of H.245 call parameters. (Also known as RTCP)	Polycom
5004 – 6004	Dynamic	TCP	There is no registered port for this application, Vcon uses an available port for H.245 (Call Parameters)	Vcon
5004 – 6004	Dynamic	UDP	There is no registered port for this application, Vcon uses an available port for Real-time Transport Protocol (RTP) for streaming video.	Vcon
5004 – 6004	Dynamic	UDP	There is no registered port for this application, Vcon uses an available port for Real-time Transport Protocol (RTP) for streaming audio.	Vcon
5004 – 6004	Dynamic	UDP	There is no registered port for this application, Vcon uses an available port for Real-time Transport Control Protocol (RTCP) for streaming video and audio.	Vcon
5555-	Dynamic	TCP	Q.931 Call setup	Tandberg

5556				
11720	**Static**	**TCP & UDP**	**This port is registered to Cisco and is used as an alternative for call set-up – h 323hostcallsigalt (must be bi-directional)**	**Polycom**
22136	**Static**	**TCP**	**There is no registered port for this application, Vcon uses an available port for remote Vcon endpoint administration**	**Vcon**
26505	**Static**	**TCP**	**There is no registered port for this application, Vcon uses an available port for Remote Console**	**Vcon**

Other Port Numbers Used in Videoconferencing

Range	Type	Protocol	Application	Man.
49152 – 49159	**Dynamic**	**UDP**	**Sony uses this range of ports for audio and video data streams**	**Sony**
49152 – 49239	**Dynamic**	**UDP**	**Sony uses this range of ports for multipoint**	**Sony**

Polycom

PORT	TYPE	PROTOCOL	DESCRIPTION
224.0.1.41:1718	Static	TCP & UDP	h323gatedisc (must be bi-directional)
1719	Static	TCP & UDP	h323gatestat Gatekeeper RAS (must be bi-directional)
1720	Static	TCP & UDP	h323hostcall Q.931 (Call Setup) (must be bi-directional)
1731	Static	TCP & UDP	msiccp Audio Call Control (VoIP)
3230 – 3247	Dynamic	UDP	Audio and Video (must be bidirectional)
3230 – 3235	Dynamic	TCP	H.245 call control: aka RTCP (must be bidirectional)
Other:			
PORT	**TYPE**	**PROTOCOL**	**DESCRIPTION**
21	Static	TCP	FTP allows upgrade of endpoint software (must be bidirectional)
23	Static	TCP	Telnet (must be bidirectional)
80	Static	TCP	Web browser interface to codec controls and menus
389	Static	TCP	ILS Registration (LDAP)
1300	Static	TCP & UDP	h323hostcsllsc H323 Host Call Secure
1503	Static	TCP & UDP	T.120 (Data Channel in a multipoint)
2979	Static	TCP & UDP	H.263 Video Streaming
11720	Static	TCP & UDP	h323callsigalt H.323 Call Signal Alternate

Sony PCS – X

PORT	TYPE	PROTOCOL	DESCRIPTION
1718	Static	TCP	h323gatedisc (must be bi-directional)
1719	Static	TCP	h323gatestat
1720	Static	TCP	H323hostcall
2253 – 2255	Dynamic	TCP	H.245(Call Parameters)
49152-49159	Dynamic	UDP (RTP/RTCP)	Audio & Video Data Streams
49152 – 49239	Dynamic	UDP	Multipoint

Tandberg

PORT	TYPE	PROTOCOL	DESCRIPTION
1719	Static	UDP	Gatekeeper RAS
1720	Static	TCP	Q.931 (Call Setup)
5555 – 5556	Dynamic	TCP	H.245(Call Parameters)
2326- 2373	Dynamic	UDP	Video Data Streams
2326- 2373	Dynamic	UDP	Audio Data Streams
2326- 2373	Dynamic	UDP	Data/FECC
21	Static	TCP	FTP
23	Static	TCP & UDP	Telnet & NTP listening socket
80	Static	TCP	HTTP
123	Static	UDP	NTP
161	Static	UDP	SNMP (Queries)
962	Static	UDP	SNMP (Traps)
963	Static	TCP	Netlog
964	Static	TCP	FTP/data
965	Static	TCP	VNC
970	Static	UDP	Streaming/RTP Video
971	Static	UDP	Streaming/RTCP Video
972	Static	UDP	Streaming/RTP Audio
973	Static	UDP	Streaming/RTCP Audio
974	Static	UDP	SAP (Stream is directed to 224.2.127.254:9875)

Vcon

PORT	TYPE	PROTOCOL	DESCRIPTION
1718	Static	UDP	h323gatedisc (must be

			bi-directional)
1719	Static	UDP	Gatekeeper RAS
1720	Static	TCP	Q.931 (Call Setup)
5004 – 6004	Dynamic	TCP	H.245(Call Parameters)
5004 – 6004	Dynamic	UDP (RTP)	Video Data Streams
5004 – 6004	Dynamic	UDP (RTP)	Audio Data Streams
5004 – 6004	Dynamic	UDP (RTCP)	Control Information
Optional:			
PORT	**TYPE**	**PROTOCOL**	**DESCRIPTION**
389	Static	TCP	ILS Registration (LDAP)
1002	Static	TCP	Site Server Registration (Windows 2000 Built-in LDAP)
1503	Static	TCP	T.120 (Data Channel)
22136	Static	TCP	VCON MXM – Remote VCON Endpoint Admin
26505	Static	TCP	VCON MXM – Remote Console

Appendix B

Polycom CMA and Polycom PVX System Requirements and Documentation

Please refer to the Polycom website for the most current documentation:

http://support.polycom.com/PolycomService/support/us/support/video/index.html
or
http://support.polycom.com/PolycomService/knowledgebase/search.htm

Web cam recommendations may become outdated. Therefore a make and model will not be specified in this document. Today's technology allows for better web conferencing with advances in web cam manufacturing. To obtain the best experience it is recommended to avoid using a web cam older than 5 years old.