POLYCOM®
Network Systems

Deployment Of Multi-Network Video And Voice Conferencing On A Single Platform

Technical White Paper

Document Overview

This document provides an overview of the issues, capabilities and benefits to be expected from a single platform solution for video and voice multipoint and gateway conferencing. How Polycom meets these challenges and how to configure the Polycom Unified Conferencing Bridge to meet your requirements are also addressed.

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TABLE OF CONTENTS

Introduction

With ever-increasing deployments of voice and video bridging infrastructure, as well as Web collaboration, endusers are demanding much more than "plain vanilla" conferencing services. Endusers now want the ability to:

- Integrate voice, video and web collaboration into the same conference
- Have simple, intuitive user interfaces
- Have ad hoc conference service support, regardless of communications media

Networks today are delivering:

- An ever-expanding range of network technologies
- An increasing range of endpoints with different communications media, network support and performance capabilities

Decisions to invest in infrastructure and conferencing technologies are getting more complex, with a wide array of equipment, software and network choices, and uncertainty about where these technologies and their endusers are headed. As a result, support personnel require help in making these decisions, and alternatives that minimize the risk of the investment decisions they are required to make.

Today's separated communications and conferencing technologies reside on different networks, on different platforms requiring different management interfaces for support. These communication boundaries are major barriers to the effective deployment of mixed / feature rich media conferencing services. A unified conferencing platform solution that supports unified conferencing services, and that can be managed with a single management interface, is required to overcome the complexities associated with these historical barriers.

To provide effective voice and video conferencing services, organizations are looking for a converged solution, a solution that will answer all the above challenges. There is a need for a unified conferencing bridge that supports multiple networks such as ISDN and IP, multiple standards such as H.320 and H.323, and multiple products and applications such as ISDN voice and video, IP voice and video, PSTN voice, and wireless (cellular) voice.

How Does The Polycom MGC Meets These Challenges?

MGC Unified Conferencing Bridge

Polycom developed the Multipoint Gateway Controller (MGC) conferencing bridge to support multipoint and gateway conferencing that seamlessly bridges conferences across different networks, as well as different endpoints with different communications media support and performance capabilities. Automatic transcoding of differences in audio and video algorithms, video resolution, video frame rates, transmission rates and network protocols are integral to the MGC architecture. Transcoding capability across these technology boundaries is an essential feature of the Polycom MGC enabling support of ISDN video, IP video, PSTN voice and VoIP conferences on a single platform.

The MGC architecture also includes the ability to share hardware and software resources across all switched and IP video, voice and mixed video / voice conferences. Key benefits of the MGC shared resource design include lower costs of equipment needed to support video and voice conferencing, improved resource module backup, and more efficient use of platform "real estate" that can be used for future system upgrades.

Polycom's MGC conferencing bridge is designed to deliver high levels of system reliability and ease of support. Hardware modules and power supplies are all accessible from the front of the MGC platform and are hot-swappable. The self-configuring design of the MGC hardware modules means that replacing a module does not require an engineer or a technician. The diagnostic and fault resolution capabilities of the MGC enable effective remote diagnostics of alarms down to the DSP level. These combined system capabilities, with the shared resource architecture of the MGC, results in high levels of system reliability and ease of support needed for an effective Unified Conferencing Bridge solution.

Supports Multiple Networks

The MGC Unified Conferencing Bridge streamlines voice and video services by providing a single entry point to multiple networks, protocols, standards and services integrated for IP, ISDN, POTS and wireless (cellular) multipoint and gateway conferencing applications. The ability to seamlessly provide video and voice communications connectivity from anywhere to anywhere across different networks on the same platform is a major benefit that should be demanded from a single conferencing platform solution. The complete and automatic transcoding capabilities of the MGC are essential to achieving this key benefit.

The resource sharing design of the Polycom MGC provides the opportunity to reduce the required investment in network resource interfaces. It also provides the opportunity to reduce the costs of network service, as the same switched or IP network connections can be used to support ISDN video, IP video, PSTN voice and VoIP conferences.

The capability of the MGC to support multiple networks on a single conferencing platform also has important benefits in migrating to new system support for additional network protocols and standards. The fact that these migrations, like ISDN video and voice to IP networks, occur over many years, is a very important reason why an effective single

conferencing bridge solution is needed to address the complexity, increased costs, and inefficiency in supporting different multipoint and gateway conferencing platforms.

The MGC capability to use the same IP resources to support video and/or voice multipoint and gateway conferences means that decisions and investments in IP are leveraged in the ability to support the migration to IP video and voice, at whatever rate it occurs.

Ease Of System And Conference Support

A key requirement and benefit of a single platform conferencing bridge is that the system and conferences are easy to support, both from a support and enduser perspective. A key component of system reliability is to have a management application that enables easy monitoring of conferencing operations, to quickly resolve any system or conference alarm.

Polycom's answer to this requirement is the MGC Manager, a Windows-based system and conference management application that enables support to manage either a single or an entire network of MGCs. From the MGC Manager application window, video and voice conferences can be scheduled and managed, conferences and systems can be monitored for alarms, and problems resolved. Hardware alarms can be diagnosed down to the DSP level, and reset or taken off-line to allow continued use of all other hardware resources for video and voice conference support.

Maintaining the MGC platform at current release levels of software is a simple download of the latest release of the MGC operating system. The MGC automatically upgrades the software level of hardware modules when a new software release is downloaded.

At the enduser level, the requirement for management tools and interfaces that are simple and easy to use is equally important. The benefits of a single conferencing platform solution are significantly reduced if the enduser must learn to use different interfaces in scheduling and managing their video and voice conferences. Different user interfaces means: more enduser training, more enduser errors that require more support and, less cross-over utilization of different conferencing media.

Polycom's Web Commander browser-based interface for endusers is a single enduser tool to schedule and manage video or voice conferences on a single or multiple network of MGC gateways, MCUs and Unified Conferencing Bridges. Having a single interface to access and manage any combination of video and voice conferences, whether the network is switched, dedicated or IP, means that meetings will be more productive and that the communications media that best addresses the enduser's conference requirements will be used.

Feature Rich

Ease and simplicity of use are not sufficient benefits to have an effective single conferencing platform solution, if it does not also support the full set of video and voice conferencing features that the enduser requires. Most video MCUs today are capable of supporting either ISDN or IP video conferences with expected conferencing features, as well as multipoint ISDN or IP voice conferences. In the latter case, the voice conferencing capabilities of these video MCUs do not include real voice conferencing features that the enduser expects. The

lack of features such as roll call, voting and polling, and IVR / DTMF interfaces are key reasons why these solutions for video and voice conferencing have not displaced any significant share of voice conferencing services or equipment sales.

The Polycom Unified Conferencing Bridge provides the enduser with fully featured video and voice conferencing capabilities that are also, in many cases, replicated across these different conferencing environments. Support for attended, unattended and ad hoc video and voice conferencing services provides the enduser and operations support with important service level consistency, and the flexibility to use a level of service that best supports the needs of the conference. Equally important in this regard is a consistent interface for the enduser to schedule and/or manage their conferences.

Polycom's Web Commander is the only browser-based interface the enduser requires to schedule or manage a video, voice or mixed conference on the Polycom Unified Conferencing Bridge. Polycom's Web Commander also supports the integration of video and voice conferences with Web-based collaboration, the Polycom Web*Office*. The integration of voice and video conferencing services with Web collaboration is now a baseline service capability that a single conferencing platform solution should address.

Integration of conferencing features and services is the direction that endusers are expecting and vendors are rapidly developing. By definition, a single conferencing platform solution should offer a rich selection of possible conferencing features, and a seamless integration of different conferencing and communications media.

The Polycom Office is the conferencing and communications vision that drives the integration of all Polycom video, voice, network and data products. Polycom's Unified Conferencing Bridge is a major advancement in making a feature-rich, seamlessly integrated conferencing environment a reality.

Flexible Deployment Options

The Polycom MGC Unified Conferencing Bridge supports the flexible deployment of multipoint and gateway conferencing capabilities, whenever and wherever they are needed. The MGC-50 (8) slot platform or a MGC-100 (16) slot platform can be selected to configure a video and/or voice multipoint, gateway, or combined multipoint and gateway configuration.

As requirements for video and voice multipoint and gateway conferencing change, upgrades to add additional video and / or voice conference capabilities can be easily accomplished with the addition of the appropriate hardware resource modules and software support. All MGCs, regardless of configuration and conference support capabilities, use the same platforms, hardware resource modules and operating system software, making it both easy and cost-effective to support a network of systems. Any or all can be easily upgraded to the full conferencing capabilities of a single platform MGC Unified Conferencing Bridge.

Return On Investment

A single multi-point and gateway conferencing platform should deliver feature-rich service level capabilities and seamless integration of media.

The most important deliverable is a very attractive return on investment. This is inherent in the purpose of having a single platform solution. In this regard, the design capability of the Polycom Unified Conferencing Platform to share software and hardware resources, to be managed with one interface, and to transcode across a full range of video, audio and network standards, means the customer can leverage their investment in equipment and save on support costs in ways that are both flexible and produce significant returns on investment.

Meeting The Challenges

Effectively deploying, managing and using integrated or converged voice and video services has been a difficult challenge in the past. Use of these services required multiple and different specialized applications, products, systems, networks, multiple endpoints with different communications capabilities, multiple access Ids, and no common model for security and/or operation.

The Polycom MGC Unified Conferencing Bridge has the design architecture to support conferencing services across diverse and separate communications networks, using this technology to make it easy. Building a MGC Unified Conference Bridge configuration requires an analysis of conferencing needs and shared resource opportunities that make sense for your conferencing and operations environment.

The following is an example system configuration that we will use to follow the tradeoff considerations made in developing an MGC Unified Conferencing Bridge system configuration.

Example Conference Support Requirements:

- 48 Ports voice conferencing
- 24 Ports IP (H.3233)
- 12 Ports ISDN (H.320)
- Video continues presence
- Conferences types:
    - VoicePlus voice conferencing
    - Ad-hoc (reservation-less) support for video conferencing
    - Mix IP and ISDN
    - Audio transcode
    - Dial through IP out to ISDN (GW sessions)
    - Dial through IDSN in to IP (GW sessions)

MGC Building Blocks

Step 1 – ISDN Network Resource Requirements

ISDN users are H.320 ISDN participants and PSTN voice participants

- **H.320 ISDN video network resource requirement**
  First, the typical rate that participants will use (128kbps/384kbps/768kbps…) must be considered. The rate selected determines both ISDN network and Multiplex (MUX) resource requirements.
  N(number of channels) = B(number of channel per call) x P(number of participants)
  In this example, 384kbps is the typical connection rate.
  N = 6 (for 384 call) x 12 (participants) = **72 B channels**
- **PSTN voice network resource requirement**
  The voice network requirement is always one B channel per participant
  N = 1 x P (number of participants)
  Therefore, N = 48 (48 voice participants) = **48 B channels**

- **Shared resources optimization**
  It now is necessary to decide how many voice and ISDN video conference participants will be supported simultaneously.

  Polycom has 3 types of ISDN network modules that support the network port capacity of the MGC being configured:
  Net 2 – supports 2 T1/E1/ISDN network connections
  Net 4 – supports 4 T1/E1/ISDN network connections
  Net 8 – supports 8 T1/E1/ISDN network connections

  One option is to support all voice and video endpoints simultaneously. In this example, 96(Video) + 48(voice) = **120 B ISDN channels** are needed.

  The other option supported by the Polycom Unified Conferencing Bridge is to share the network resources to support both ISDN video and PSTN voice from one pool of network resources.

  In this example, the logical range of capacity scenarios includes the following:

	ISDN VIDEO (@384KBPS)	VOICE
Scenario 1	15 endpoints / 90 B channels	2 endpoints / 2 B channels
Scenario 2	7 endpoints / 42 B channels	48 endpoints / 48 B channels
Scenario 3	8 endpoints / 48 B channels	44 endpoints / 44 B channels

  In this example, we will utilize the NET-4 module, that providing *92* B channels (T1).

Step 2 – Audio Resource Requirements

- **H.320 ISDN voice resource requirement**
  Each participant requires one voice resource port.
  Therefore, support for 12 ISDN H.320 video conference participants requires **12** voice resource ports.
- **PSTN voice resource requirement**
  Each participant requires one voice resource port.
  Therefore, 48 PSTN voice conference participants requires **48** voice resource ports.
- **H.323 IP Voice Resource requirement**
  Each participant requires one voice resource port.
  Therefore, 24 H.323 IP video conference participants requires **24** voice resource ports.
- **Shared resources optimization**
  Next, decide how many voice, ISDN video and H.323 IP video conference participants will be supported simultaneously.

  Polycom has 2 types of audio modules to support the voice port capacity of the MGC being configured:
  Audio+ 48 – supports 48 voice conference participants and/or 24 video conference participants
  Audio+ 24 – supports 24 voice or video conference participants

  One option is to have the capability to support all PSTN voice, ISDN video and H.323 IP video endpoints simultaneously. This will require 12 (ISDN) 48 (PSTN) + 24(IP) = **84** voice ports (i.e. 1 **Audio+ 48** and 2 **Audio+24** modules = 3 modules in total).

  The other option supported by the Polycom Unified Conferencing Bridge is to share the network resources to support PSTN voice, ISDN video and H.323 IP video conference participants from one pool of voice resources.

  For our example the logical range of capacity scenarios includes the following

	VIDEO (ISDN)	VOICE (PSTN)	VIDEO (IP)	TOTAL
Scenario 1	12	0	12	24(V)
Scenario 2	0	0	24	24(V)
Scenario 3	0	48	0	48(A)
Scenario 4	12	24	0	12(V)+24(A)
Scenario 5	0	24	12	12(V)+24(A)
Scenario 6	6	24	6	12(V)+24(A)

  In this example, we will utilize one **Audio+ 48** module.

Step 3 – IP Resource Requirements

Each H.323 IP video conference participant requires one IP module resource port.
For our example 24 IP H.3233 ports require the **IP 48 / 24** module.

Step 4 – MUX Resource Requirements

First we need to decide what is the typical rate that ISDN participants will use (384Kbps/768Kbps…) The selected optimized rate determines the **MUX** module capabilities that are required.

N (number of **MUX** modules) = P (number of participants) / O (module capacity)

O- Module capacity Vs. rate

RATE	**384**	**768**	**E1/T1**
Capacity	16	8	4

In this example, the optimized rate selected is 384kbps
N = 12 (participants) / 16 (optimized to 384 kbps) = 1 MUX resource module

Step 5 – Video Continuous Presence & Transcoding (CP/TX) Resource Requirements

The Polycom MGC CP/TX resource module supports both enhanced continuous presence capability for 21 different CP layouts as well as transcoding of endpoint equipment differences in audio, video, data and transmission rates.

Each video conference participant requires one video CP/TX resource port.

Polycom has 2 types of CP/TX modules to support the CP port capacity of the MGC being configured:

CP/TX 6 – supports 6 video conference participants in a CP supported conference
CP/TX 12 – supports 12 video conference participants in a CP supported conference

One option is to have the capability to support all H.323 IP video and ISDN video endpoints simultaneously. In this case, 12 (ISDN) and 24 (IP) = 36 video ports (6 modules) are required.

CP/TX conference support resources have a cost in dollars as well as in chassis slots. From a conference applications requirement perspective, the use of continuous presence does not always enhance the conference.

For these reasons, it is most often decided in configuring a MGC to share the CP/TX resources to support ISDN video and H.323 IP video conference participants from one pool of

resources. The following shows what the resource requirements are to support hardware-assisted continuous presence and transcoding on the MGC at different capacities.

CP/TX PORTS	**6**	**12**	**18**	**24**
CP/TX Modules	1	2	3	4

In this example, we will utilize one video double module (12 Video ports).

Step 6 – MGC 50 / 100 Platform Selection

Polycom MGC system configurations can be built on either the MGC-50 or the MGC-100 platforms. All of the key system attributes of shared resources - transcoding, operating system, and management interfaces - are identical in the MGC-50 and the MGC-100. There are some important differences in the chassis architectures that should be considered in making a MGC platform decision.

The following summarizes these differences:

	MGC-50	**MGC-100**
Number of Slots	8	16
Hot Swappable Power Supplies*	NO	YES
Width	19”	21”

*In both the MGC-50 and the MGC-100, all functional modules are hot swappable

Beyond these differences in the MGC-50 and the MGC-100 platforms, the only other key considerations are the number of resource modules that are required to build the unified conferencing system today as well as what is expected in terms of future expansion of capacities and conferencing features.

The following is a summary of the resources and individual resource capacities that have been identified as needed to support the example of a single (shared resource) conferencing solution.

Module	**#**	**CAPACITY**		
		Voice	**Video**	**Remarks**
Net-4 *	1	92	15	Shared for Voice (PSTN) and H.320 participants
MUX	1	NA	16	ISDN (H.320) participants only
IP48/24	1	NA	24	IP participants only
Audio+ 48	1	**48**	**24**	Shared for Voice and Video (IP/ISDN)
Video	2	NA	12	Shared for H.323 and H.320 participants
TOTAL	**6**			

*In this example, the network is assumed to be T1 / PRI.

For the above configuration, we can choose the MGC 50 or the MGC 100 platform. An MGC 50 platform configuration of the example requirements will have 2 empty resource slots for future expansion. A MGC 100 configuration has 10 empty resource slots for expansion, as well as the capability to be configured with redundant power supplies. If expansion slots and power redundancy are important requirements, then the MGC 100 is the platform to select. If additional resource capacity in any category is required, then this also might make the MGC 100 the better platform decision.

Because the MGC 50 and the MGC 100 are built on the same system architecture that supports all of the multipoint and gateway conferencing scenarios discussed, MGC 50 systems are easily upgraded to the MGC 100 platform using the same resource modules. Any MGC 50 or MGC 100 can also be easily reconfigured to support any change in systems deployment that may occur in the future.

Step 7 – Upgrade Options / Considerations

We can see in the above table that the network resources built into the configuration have more available capacity than the capacity built to support audio ports. This points to at least one more consideration in configuring the system today or for a future upgrade path.

Because the network interface capacity is already built into the system configuration, the addition of one more **Audio 48** resource module can provide the following capabilities:

	VIDEO (ISDN)	VOICE (PSTN)	VIDEO (IP)	TOTAL
Scenario 1	7	48	17	24(V)+48(A)
Scenario 2	0	48	24	24(V)+48(A)
Scenario 3	0	92*	0	92(A)
Scenario 4	14	8	24	42(V)+8(A)

* ISDN Network limitation

This potential upgrade option is a good example of the costs savings that can be expected from a Unified Conferencing Bridge. For the cost of 1 **Audio 48** port resource module, this system's capabilities are increased as follows: voice conferencing capacity is increased from 48 to 92 ports and mixed support capacity for video and voice is increased from 36 to 72 ports.

Step 8 – Supported Conference Types

Examples of Conference Types Supported:

- Mixed IP (H.323), voice (PSTN) and ISDN (H.320) participants all in one conference.
- Audio transcoding inherent in every port
- Dial through IP out to ISDN (GW sessions)
- Dial through IDSN in to IP (GW sessions)
- Dial through IP to IP (GW sessions)

Voice*Plus*

Our example definition includes support for 48 ports of Voice*Plus* conferencing. Voice participants will be able to enjoy the following features :

- IVR service messages
- DTMF detection
- Roll-Call announcements
- DTMF conference control – Mute, lock, help, invite
- Voting
- Polling
- Automatic Gain Control

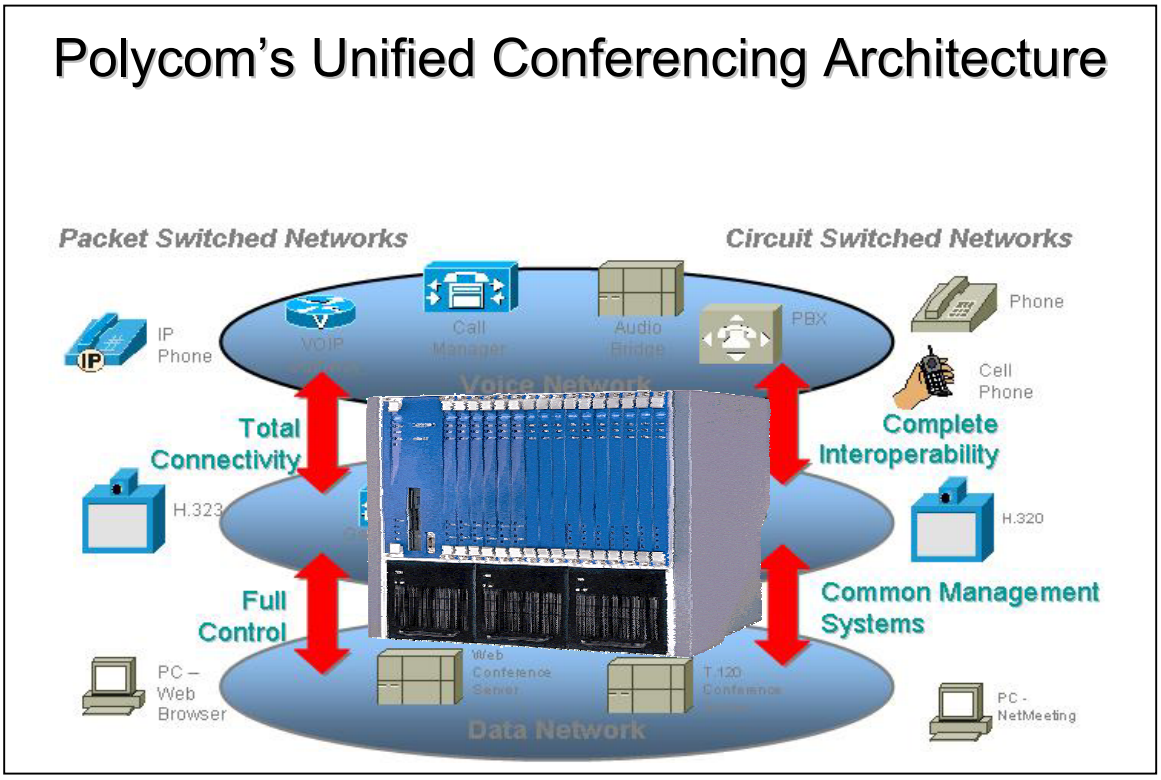
To enable this functionality, we need to add support for the Voice*Plus* IVR/DTMF software to the example system configuration.

Ad-hoc (reservation-less) Conference Service

Support for ad hoc conference service support is provided by Polycom's Virtual Conference Suite of conferencing features. Key features supported include: single phone number per conference, virtual conference meeting rooms, add new conference participants, auto extend, and auto terminate the conference.

Summary: MGC Unified Conferencing Bridge Benefits

The power to bring together integrated services for voice and video into one platform is beneficial on many levels. The following graphic depicts the ability to bind all layers through the MGC unified platform and provide seamless connectivity, management, interoperability, control and a simplified user experience for the network manager, the conference administrator and the enduser.



Binding these network layers, protocols and applications together under a single, unified platform has significant benefit. Let's take another look at the challenges discussed earlier and to see how the MGC provides valuable solutions to those challenges.

Beginning with the **network**, it becomes clear that the MGC platform solves many issues related to the network. The MGC provides physical interfaces to both voice and video circuits. It is extremely important to have a common element when deploying mixed services. Having a system that provides mixed services natively, within a reliable architecture, alleviates many problems associated with simply getting the services deployed.

The MGC platform supports all of the various **Standards and Protocols** associated with voice and video services. This is important in itself, but the MGC goes a crucial step further.

The MGC solves the issue of mixing conferences that have differing standards and protocols. NO LONGER are you limited to supporting a least common denominator method of conferencing services. The MGC translates the various standards and protocols between each endpoint, allowing each endpoint to participate in a conference session at its highest mode of operation. This is achieved entirely without manager or enduser intervention.

The **Network Manager** now has a common entry point for monitoring, diagnostics and management. It is much easier for the manager to have access to and manage one or more systems using the robust capabilities of the MGC Manager, ensuring with confidence that all voice and video services are operational from end to end. The other key element of benefit to the manager is that the MGC unified platform is based on the highly reliable MGC architecture, providing scalability and redundancy.

The **Conference Administrator** also now has a single point of entry for all scheduled or adhoc conferencing services. This is very advantageous to the administrator when trying to coordinate not only voice and video services, but also peoples' schedules.

The Enduser now has a single point of entry for conferences and a common user experience for voice and video services. The enduser is able to focus on the issues at hand without having to worry about using the technology.

The MGC eliminates issues related to **differing levels of access control and security.** As described earlier, the MGC provides common entry points into the service based on a user's, or group's policies and permissions. The network administrator sets these policies and permissions one time and, from that point on, the user's experience remains the same.

The organization's typical usage scenarios and requirements are now greatly expanded due to the ability to mix and integrate both on-demand and scheduled voice and video services.

The organization's goals for technical support, effective management of network, technology, and application resources are now all synchronized through a core set of capabilities based on the MGC Unified Conferencing Bridge.

All of these benefits result in a lower cost of ownership and operation, thereby providing cost-effective services to the organization.