TANDBERG MPS J3 Release Document

TANDBERG

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1. Document Revision History

Rev 1.5	- Included Release of J3.3 Minor Release
Rev 1.4	- Included Release of J3.2 Minor Release
Rev 1.3	- Included Release of J3.1 Minor Release
Rev 1.2	- Corrected error in Flexible Resource Allocation capacity table
Rev 1.1	- Corrected Known Limitations
Rev 1.0	- Included Release of J3.0, Initial Version

2. Release Notes for the TANDBERG MPS Software Version J3.3

2.1 Introduction

These release notes describe the new features and capabilities included in the TANDBERG MPS software version J3.3 released on 24 January 2007.

2.2 New Feature Overview

2.2.1 V.35

2.2.1.1 Restricted Rates on V.35 Clock Generation

It is now possible to choose restricted rates when setting the Internal Clock Rate within the Serial V.35 configuration. This allows 56k increments within the clock rate in addition to the 64k increments previously supported. To use this functionality simply set the Internal Clock Rate to the desired rate (in 64k increments) and then set Restrict to On. When set to ‘Off’, 64k increments will be used.

2.2.2 API

2.2.2.1 Internal Bandwidth Restrict Setting

With the inclusion of the new V.35 clock rate restriction a new API command has been added. This is to allow the setting of restricted clock rates to be set through the API as opposed to the Web interface of the MPS. To use this feature the following API commands are used. When set to Off 64k rates are used; when set to On 56k rates are used. Replace the “1” in the below command to the correct V35 card for your application.

Command:

xConfiguration SerialInterfaceCard 1 InternalBandwidthRestrict: <On/Off>

2.2.3 Audio

2.2.3.1 Disable H.245 DTMF

It is now possible to disable out of band DTMF tones so audible in band DTMF tones can always be used if needed. This is useful for when H323 sites connected to the MPS need to send DTMF tones to other connected sites. To use this setting the following API commands are used.

Command:

xConfiguration SystemUnit OutbandDTMF: <On/Off>

2.3 Changes and Improvements since previous version

2.3.1 ISDN

- Resolved an issue where ISDN bit errors could cause an unexpected restart
- Resolved an issue where the MPS could reject incoming ISDN calls on certain channels
- The system will no longer hang when trying to use an invalid Network ID when placing ISDN calls
- Resolved an issue where an unexpected restart could occur if disconnecting an ISDN participant while changing CP layouts

2.3.2 V.35

- Improved sync loss timer so that it will not trigger the password timer which may have caused ports to disconnect from conferences
- Corrected an issue that could have caused a V.35 firmware upgrade to fail if upgrading a V.35 card inserted into slot 1

2.3.3 Web Management

- Corrected an issue where the wrong participant name could be shown on the conference overview page for certain sites
- Resolved an issue where the disconnect site button may not work for H.323 telephony sites
- The SIP Configuration page is now displayed in English
- Improved configuration changes to eliminate possible restarts when pressing Save button
- Corrected issue where the main video snapshot could be displayed as the dual stream snapshot
- Added clock transmit (TX) and receive (RX) sync status to the Serial V.35 status page
- Changed all labeling of ‘Duo Video’ to ‘Dual Video Stream’
- Added redial button on conference overview page for V.35 sites
- Web snapshots are no longer accessible when disabled
- The PRI status on the main overview page will now show in red if any enabled interface that is not active

2.3.4 Audio

- Corrected an issue where a sudden burst of audio RTP packets could cause an unexpected restart
- Corrected an issue where ISDN telephone sites could lose audio during a call
- ISDN bit errors in the audio stream will no longer cause an unexpected restart

2.3.5 H.323

- Corrected issue where H.323 sites could remain in the clear out state
- Improved FUR (Fast Update Request) handling when in oVS mode to eliminate possibility of an unexpected restart
- Corrected an issue where H.323 calls utilizing network ID 2 would fail to connect when using the Assent protocol
- The MPS will no longer restart when trying to send RAS messaged to an invalid IP address
- Corrected an issue where severe packet loss may have caused an unexpected restart

2.3.6 System

- Improved event log system to update the system controller logs during all restarts
- The system will no longer use the default Password Reject slide when a custom slide has been uploaded to the system
- Removed all encryption capabilities from the non-AES software file

2.3.7 SIP

- Entering a domain name that is too large will no longer cause an unexpected restart

2.3.8 Video

- Improved video transition from CP layouts to oVS mode to eliminate poor video
- Improved video mute function while in CP calls with Conference Selfview set to Off to eliminate incorrect site muting
- The MPS will now voice switch properly when a site disconnects and reconnects while all other participants are muted
- The Participant Identifier Timeout setting now works correctly
- Corrected an issue that may have caused poor video when using wide CP layouts
- Improved the Layout Locking feature when used in conjunction with Force Floor
- The MPS will no longer remove Wide CP layouts when a site has downsped to the QCIF encoder
- The MPS will no longer attempt to send CIF from the QCIF encoder which may have caused an unexpected restart
- The MPS will no longer toggle between 4CIF and CIF when in a CP call and a site requests the floor
- ISDN bit errors in the video stream will no longer cause an unexpected restart
- Corrected issue where the MPS could display incorrect video text
- The MPS will no longer change video format to CIF due to packet loss downspeeding

2.3.9 API

- The MPS will no longer display “sh: /usr/bin/sendtrap: not found” when login fails through Telnet

2.3.10 XML

- The MPS will no longer restart when posting a malformed XML document

2.3.11 Gateway

- Removed Load Limit setting from the Gateway Configuration when the unit is a combination MCU/GW

2.3.12 Security

- Telephone calls will now be disconnected when attempting to join an encrypted ad hoc conference

2.3.13 Inter-Op

- The MPS will now correctly decode H.261 video from a Polycom MGC when using AES encryption even though the payload type is incorrect

2.4 Known Limitations

- Moving an ISDN participant between conferences will fail unless Welcome Picture and Sound is set to Off. This will be resolved in a future software release.
- TANDBERG Classic E/B series codecs may not handle video transitions well when using oVS in the MPS. This issue will be resolved in a future endpoint software release.
- Duo Video is not supported in a high resolution oVS conference. H.239 however is fully supported.

3. Release Notes for the TANDBERG MPS Software Version J3.2

3.1 Introduction

These release notes describe the new features and capabilities included in the TANDBERG MPS software version J3.2 released on 18 July 2006.

3.2 New Feature Overview

3.2.1 Video

3.2.1.1 Optimal Voice Switching (oVS)

J3.2 includes a new feature for improved video quality as well as support for Optimal Definition (400p/448p) and HD (w720p). When enabled and in a Voice Switching, IP-only conference, the MPS will pass all video through, untouched, as long as all endpoints share a similar capabilities set. This allows the MPS to host high resolution conferences while minimizing any latency and video degradation that would be caused by any video processing performed by the MPS. To ensure the success of all calls, regardless of the connected participants, transcoding may be initiated if the MPS determines that it is required to maintain connectivity to all parties. This allows meetings to continue uninterrupted regardless of endpoints connected. This setting is available on a per-conference configuration and can be enabled/disabled while calls are active.

Note: H.320 endpoints will always receive transcoded video.

Note: Site naming and the telephone icon are not available in this mode as the MPS does not process the video.

3.2.1.2 Dual Stream Switching

Similar to the new oVS feature, the MPS now supports Dual Stream Switching. This allows dual streams (both DuoVideo and H.239) in IP-only conferences to pass through the MPS untouched unless transcoding is necessary, preserving image quality and minimizing video latency. Additionally, this mode allows for 2 individual video rates to be transcoded while dual streams are active. To ensure the success of all calls, regardless of the connected participants, transcoding may be initiated if the MPS determines that it is required to maintain connectivity to all parties. When transcoding the dual streams is necessary, the MPS will use 1 video encoder for the dual stream and 1 for the main conference stream like previous software versions. This feature is always enabled and can only be disabled by setting the ‘`Legacy Level`’ to a value of 1-15 for the conference.

Note: H.320 endpoints will always receive transcoded dual streams.

3.2.1.3 Encoder Selection Policy

Included in J3.2 is the ability to set the transcoding preference for the conference. In previous software versions the MPS would only engage transcoding when sites were connected at different call rates, regardless of the video algorithms used. This mechanism has now been modified to include video algorithm as a preference. Additionally, fallback mechanisms have been implemented to ensure the best quality video in all situations.

When configured to '`Best Bit Rate`', the MPS will now transcode different video algorithms (one high and one low) if all endpoints are connected at the same video rate. If one site connects at a different bit rate or downspeeds to a lower bit rate, the MPS will then function as it did in previous versions.

If set to '`Best Video Standard`', the MPS will then transcode based on the video algorithms sent to each site, regardless of the speed in which the site is connected. If all sites are capable of receiving the same video standard, the MPS will transcode based on the call speed in which endpoints are connected.

3.2.2 Management

3.2.2.1 Disable Web Snapshots

It is now possible to disable Web Snapshots at a per-conference level. This allows for greater security for conference monitoring when used in conjunction with TMS (TANDBERG Management Suite). This setting is available on conference configuration and can be enabled/disabled while calls are active.

3.2.2.2 Bandwidth Management

To prevent individual sites from compromising the quality of an entire conference, the MPS can now move them automatically to the QCIF encoder, thereby ensuring transcoding resources are used for other sites. When configured for '`Auto`', the MPS will automatically move sites to the QCIF encoder that have downsped below the lowest rate of the call if transcoding is already active. In addition, sites that downspeed below the '`Minimum Conference Threshold`' will be moved to the QCIF encoder without any administrative interaction. '`Bandwidth Management`' may also be configured for '`Manual`', allowing the site to downspeed, possibly adversely affecting conference quality.

3.2.2.3 Move to QCIF Action Button

With the new '`Bandwidth Management`' feature, a new action button has been added to the MPS conference web interface. This new '`Move To QCIF`' button is available on the '`Change`' tab on the conference monitoring page, allowing an administrator to force sites to the QCIF encoder to prevent them from adversely affecting the conference.

3.2.3 Conference

3.2.3.1 Conference Connection Based on Caller ID

With the introduction of J3.0, the MPS allowed the use of caller ID information to allow access to active conferences. This feature has now been expanded to allow systems to call into the MPS by dialing the MPS controller card IP address and automatically connect to the correct conference. Using this feature requires use of '`Protect`' within the conference configuration to allow only the configured caller ID to connect to the conference. This feature also requires that the default IP conference is set to any legal value other than 0 (reject). When configured in this manner all calls that were dialed to the MPS Controller Card IP address will default to this configured conference, unless a conference is configured to match that specific caller ID.

3.2.4 V35

3.2.4.1 Clock Generation

If an endpoint is connected directly to the MPS without going through a real network, it is necessary to provide that system clocking. The MPS can now act as that clock source by replicating incoming clock on one of the ports of the network card to any of the other 31, thereby providing clock to the external system connected to that port. To set the clock speed, simply change the Internal Bandwidth to the desired speed on the V35 configuration page; this clock can then be set to a specific port by configuring the desired ports for Internal for Clocking on the V35 configuration page.

3.3 Changes and Improvements since previous version

3.3.1 Web

- Resolved an issue where the redial action button would redial the site only using the default Network ID and Network Module, even if a specific ID and Module were specified during the initial dial attempt.
- A site will now be disconnected when the '`Disconnect`' action button is clicked, regardless of the connection state of the site.
- The G.703 channels are now reported from the correct module on the '`Overview`' page.
- Resolved an issue where the Network Module and Network ID fields were displayed in reverse on the Phonebook page.

3.3.2 Conference

- Voice switching will no longer stop after a participant has requested and released the floor.

3.3.3 XML

- The MPS will no longer report the wrong call reference and log tag in an XML dial response.
- A boot event will no longer be sent upon initial configuration of the '`External Manager`' address.

3.3.4 Management

- Resolved an issue where moving a participant with the video muted would un-mute the video.
- The MPS will no longer take a long time to boot up when the Ethernet speed is configured for Auto, but no link is present.
- SIP reporting within a syslog has been expanded.
- V.35 site names are now displayed correctly.
- Increased maximum number of simultaneous Telnet sessions to 48.

3.3.5 API

- Resolved an issue where the layout locking window numbers did not match the numbers used on the Web interface.

3.3.6 Inter-Op

- The Polycom SP and MP endpoints can now connect to Single Number Dial-In.

3.3.7 H.323

- The MPS will now report the correct bandwidth in all Bandwidth Requests (BRQs) sent to a gatekeeper.

3.3.8 Video

- Resolved an issue where the MPS would report transmitting H.239 to a Polycom IPower when in fact it was not.
- H.239 will no longer be sent to endpoints connected via H.221 at a call rate of 2x64.
- In scenarios where the MPS is experiencing very high packet loss, the system will no longer toggle between video protocols.
- The telephone icon will now be properly displayed in all layouts. Previously, some layouts would cause the telephone icon to disappear.

3.3.9 ISDN

- If the D channel is lost due to a network failure, it will now re-sync automatically; requiring no further action from an administrator.

3.3.10 Gateway

- V.35 calls are now possible when no ISDN circuits are present on the MPS.
- The MPS Gateway will now send a Resource Available Indicator (RAI) to signal available resources to the gatekeeper. However, this will only be sent if the MPS is configured as a Gateway only.

3.3.11 V.35

- V.35 calls connected at a call rate of 384k will no longer report at a call rate of H0.

3.3.12 Ad-Hoc Conferencing

- Resolved an issue where dialing into the Single Number Dial In as a video call and the conference template has 0 video participants, the conference would start but it was not possible to end it.
- Conferences can now be created using the Single Number Dial-in via a telephone.
- If a personal conference is configured for both a create password and an access password, audio would previously fail. This has been resolved.
- Dialing an invalid Personal Conference number will no longer route to conference 1.

3.3.13 Audio

- Resolved an issue where dialing a telephone site using a DTMF string would connect the site to the conference but no audio would be heard.

3.3.14 SIP

- SIP calls will now clear out properly when disconnected; no longer getting stuck in the “clear out” state.
- Deleted SIP URI’s will no longer stay registered to the SIP server.

3.3.15 System

- If the gatekeeper settings are configured improperly, the system will no longer experience any instability.

4. Release Notes for the TANDBERG MPS Software Version J3.1

4.1 Introduction

This release note is to describe the new features and capabilities included in the TANDBERG MPS software version J3.1 released on 4/21/2006.

4.2 New Feature Overview

4.2.1 V.35

4.2.1.1 Terminal Sync Loss Timer

J3.1 includes a new feature for resetting a dial in V.35 port where the site has disconnected. This mechanism monitors H.221 framing on all connected V.35 ports. If H.221 framing is lost on any connected port a timer will engage. When the time limit has been reached the port will remain in the active state ready to accept new calls but the CP layout and conference password (if used) will be reset. As a new site connects to this port, the MPS will then prompt for the conference password again (if used) and update the CP layout if needed. The default timer value is 15 seconds to allow sufficient time for crypto resync in military installations.

4.2.2 API

4.2.2.1 Terminal Sync Loss Timer

With the addition of the new Terminal Sync Loss Timer, a new API command has been added to allow configuration of the timer value. This command can be entered through Telnet or the serial port on the MPS.

xConfiguration SystemUnit TerminalSyncLossTimer: <5..90>
Please note that the default value for this setting is 15 and all values are in seconds.

4.3 Changes and Improvements since previous version

4.3.1 Web

- Improved web server stability to eliminate possible unresponsiveness. Situation may arise when TMS is managing several simultaneous conferences while sending many conference status requests.
- Resolved issue where G.703 channels were not reported on the overview page. These are now reported as ISDN channels within the Usage graph.
- Resolved issue where it was possible to define the Default IP Net ID within the conference template and conference configuration page to a value between 1 and 32. Now only the valid values of 1 and 2 are possible.
- Resolved issue where connected V.35 calls would be reported on the overview page as used ISDN channels.
- Resolved issue where a SIP call would report a video rate that was equal to the call rate as opposed to the actual video rate being used.
- Resolved issue where the participant reconnect action button on the conference monitoring page would not trigger a page refresh when selected.

4.3.2 Conference

- Resolved issue where sites would connect slower than normal when dialing out with encryption enabled.
- Resolved issue where an unexpected restart could occur when trying to create a conference via Single Number Dial In or Personal Conference when there are no available conference resources.
- Resolved issue where an unexpected restart could occur when moving a cascade link or a site transmitting Duo Video/H.239 between conferences. These conference moves will now be rejected by the system if attempted.
- Resolved issue where an ISDN site would show as Disconnected with a cause code of 16 if added when there where no available ISDN resources. The system will now show an error message and will not allow the site to be added to the call.

4.3.3 XML

- Resolved issue where the system could display illegal characters for a site name within the status.xml document.
- Resolved issue where the MPS would delay sending parts of an XML document until it had received an ACK from the first packet of the document.
- Resolved issue where the MPS would not send feedback if the video text for a site was changed.

4.3.4 Management

- Resolved issue where setting the ethernet setting of the controller to anything other that Auto would result in the unit still using Auto Negotiation. This would lead to a duplex mismatch if the switch was not set for Auto Negotiation.

4.3.5 API

- Resolved issue where starting a conference using the API command “xcommand conferencestart” would not start the conference with the correct conference template settings.

4.3.6 Cascading

- Resolved issue where an H.243 message overflow could occur during a cascaded conference.

4.3.7 Chair Control

- Resolved issue where disabling chair control during an active conference would still allow chair control to be used.

4.3.8 Inter-Op

- Resolved issue where a Polycom PVX would stop transmitting video when joining a conference through the Single Number Dial In with H.239 enabled.

4.3.9 H.323

- Resolved issue where the MPS would not return a “User Busy” cause code when calling into a conference with no available ports.

4.3.10 Video

- Resolved issue where freezing video could occur if the endpoints where sending 4CIF/4SIF video and the MPS conference was configured with Conference Selfview Off.

4.3.11 ISDN

- Resolved issue where an unexpected restart could occur when receiving an ISDN call and the MPS was configured with single digit numbers within the PRI Configuration Number Range Start and Number Range Stop.

4.3.12 Gateway

- Resolved issue where outbound H.323 to H.320 telephone calls would be placed as 64k Data calls when using a service prefix with a bandwidth set for something other than Telephone.

5. Release Notes for the TANDBERG MPS Software Version J3.0

5.1 Introduction

This release note is to describe the new features and capabilities included in the TANDBERG MPS software version J3.0 released on 2/22/2006.

5.2 New Feature Overview

5.2.1 Ad-Hoc Conferencing

The TANDBERG MPS now supports Ad-Hoc conferencing. This allows users to create conferences by dialing predefined dial in numbers without the aid of administrators or a scheduling system.

5.2.1.1 Personal Conferences

This new Ad-Hoc feature allows an administrator to define up to 500 conferences. Each of these conferences can have a unique name, E164 Alias, H323ID, SIP URI and ISDN dial in number. This allows a user to dial their predefined conference number to create the conference which will then allow others users to dial the same number to join the conference. Optionally a create password can be assigned for greater security. An access password may also be enabled/disabled per conference. You may also define a conference template that will be used when the conference is started. This allows you to choose from one of the 10 new conference templates to customize each of these to suit the user's needs.

5.2.1.2 Single Number Dial In

This new Ad-Hoc feature allows users to dial into one predefined number and create or join conferences. This predefined number can be one or all of the following: E164 Alias, H323ID, SIP URI, ISDN number and IP Address. Users dialing into these numbers will be presented with a customizable slide and sound prompting the user to join or create a conference using DTMF tones. Users requesting to join a conference that has not yet started will be placed into a waiting room until the conference is created. Upon conference creation, all users in the waiting room will automatically be moved into the conference. Optionally a create password can be required for greater security. Please note that only conferences created using the single number dial in are accessible to users attempting to join a conference via the single number dial in. There is no access to personal conferences or scheduled conferences.

5.2.2 Conferencing

5.2.2.1 Chair Control

The TANDBERG MPS now supports H.243 Chair Control for both H.320 and H.323. This allows capable endpoints to request and release chair capabilities. While granted chair capabilities by the MPS the endpoint may perform the following actions: Assign Floor To Participant, Release Floor From Participant, Disconnect Participant, Terminate Meeting and Release Chair. Please note that only one endpoint may hold the chair at any given time during a conference. During this time the MPS will reject chair requests from other endpoints until the current chair holder releases chair.

5.2.2.2 Move Participants Between Conferences

With J3 an administrator now has the ability to move connected participants between active conferences. Using the Web interface of the MPS or API commands each connected site in a conference can be moved to any active conference. Upon entering the new conference the MPS will renegotiate the endpoint to the new conference capabilities. This includes call rate whereas the MPS will downspeed the site if it is connected at a higher call rate than the new conference allows. Please note that the MPS will not up-speed a participant to a higher call rate. If a site is moved to a conference in which the rate is higher than the site is connected, the original sites connection rate is used.

5.2.2.3 Multiple Conference Dial In Numbers

It is now possible to dynamically add extra dial in numbers to an active conference. These numbers can be ISDN, E164 Alias, H323ID and SIP URI. This allows users from different networks to access an ongoing conference or can be used to assign unique dial in numbers per participant.

5.2.3 Gateway

5.2.3.1 MPS Gateway Option

The MPS now supports H.320 to H.323 gateway functionality. The unit can be configured as a gateway only or an MCU and gateway together. This optional feature allows H.323 sites to connect to H.320 (ISDN PRI, G.703 and Serial V.35/RS449/RS530) sites. Options for the gateway are sold in groups of 10 calls with support for a maximum of 20 calls on an MPS200 and 80 calls on an MPS800. The MPS Gateway supports all of the features of the TANDBERG Gateway version G3.0 with the addition of the AAD-LD audio algorithm.

5.2.4 Video

5.2.4.1 New Continuous Presence Layouts

The TANDBERG MPS now supports 23 different screen layouts. All layout choices from J2 are still present along with the addition of 10 new 4:3 screen layouts. Also included are 8 new 16:9 widescreen layout choices. With these new widescreen layouts the MPS now supports the sending of widescreen video formats. Depending on conference configuration, 3 possibilities exist. If the conference is defined as motion, the MPS will send w288p when a widescreen layout is chosen. If the conference is defined as auto or sharpness, w576p will be sent. If the conference is configured with w720p enabled and set for auto or sharpness, w720p will be sent. Please note that all endpoints connected must support these resolutions otherwise a standard layout will automatically be chosen and sent as either CIF or 4CIF.

5.2.4.2 Layout Locking

It is now possible to lock participants within a quadrant of a CP layout. This is used to ensure that particular sites are not voice switched out of the CP image but rather remain in a fixed quadrant. This feature is administered via the Web interface. Please note that the Advanced Video Option (AVO) is required for Continuous Presence otherwise only Voice Switching is possible.

5.2.4.3 Multi Language Site Naming

The TANDBERG MPS now supports on screen site naming transmitted within the video stream. This feature can be set to Auto/On/Off per conference and when On will display the site names for each site being displayed. When set to Auto the text will be displayed for a user configurable time and will then disappear. The text will then re-appear only during a voice switching change. When set to Off no text will be displayed during the conference. This feature uses H.243 unicode which allows the MPS to display these names in their original language.

5.2.4.4 Active Speaker Indication

The TANDBERG MPS now supports an active speaker indication. This feature will outline the quadrant where the active speaker is located using a light blue border. When the active speaker changes, the border will outline the new active speaker. This feature can be enabled/disabled per conference.

5.2.4.5 Telephone Site Indication

With J3 the MPS will now encode a blue telephone icon in the upper left corner into the sent video stream. This icon indicates that there are telephone calls connected within the conference. This icon changes depending on the number of connected telephones sites and will indicate 1, 2, 3 or 3+ telephone sites connected. Also this icon will change to green when a telephone site is the active speaker. This feature can be enabled/disabled per conference.

5.2.4.6 Conference Selfview

It is now possible to remove each sites' selfview from the outgoing video stream from the MPS. When using this feature the received video on each endpoint will only contain the remote sites' video. This feature can be enabled/disabled per conference. Please note that when using this feature w720p and 2 rate transcoding will be disabled.

5.2.4.7 Low Bandwidth/3G Video Encoder

Each MPS conference now has an extra video encoder designed to handle low bandwidth and/or 3G video connections. This encoder will transmit a QCIF video stream without affecting the video stream being sent to all other participants. This encoder will automatically be chosen when the system detects a 3G participant or if a participant connects to the conference using a call rate below the minimum bandwidth threshold defined for the conference. Furthermore, the sites using this encoder will receive an optimized CP layout intended for small screen endpoints such as 3G handsets.

5.2.4.8 Minimum Bandwidth Threshold

A new conference setting has been added where a minimum bandwidth threshold can be defined. This setting defines the lowest call rate a site can connect at and still receive normal conference video. This preserves the quality of the conference by forcing all calls below this threshold to use the 3G video encoder.

5.2.5 Network

5.2.5.1 G.703

The TANDBERG MPS now supports G.703 leased line E1/T1. It is possible to define up to 5 calls per interface. This support uses the existing ISDN PRI Interface Card. Within the MPS configuration this module would be defined as a G703 card as opposed to a PRI card.

5.2.5.2 NFAS

The Tandberg MPS now supports NFAS (Non-Facility Associated Signaling). This allows an ISDN PRI D channel to be shared with other ISDN PRI circuits freeing up those channels to be used as B channels. Please note that each ISDN Interface Card can support up to 4 NFAS groups yet an NFAS group cannot span ISDN Interface Cards. This will allow up to a maximum of 8 PRI's in one NFAS group per card to be used. Please also note that backup D channel is not supported.

5.2.5.3 SIP

The TANDBERG MPS now supports the Session Initiation Protocol. This allows the MPS to now connect to SIP based endpoints such as Microsoft Office Communicator using Microsoft LCS.

5.2.5.4 Expressway Firewall Traversal

The TANDBERG MPS now supports Expressway. This allows an MPS to register directly to a TANDBERG Border Controller to achieve firewall traversal. In addition this allows the MPS to utilize the Media Direct feature of the TANDBERG Gatekeeper as well and provides flexibility in deployment.

5.2.6 Audio

5.2.6.1 Telephone Echo Suppression

The TANDBERG MPS now has the ability to suppress echo that can be generated by telephone participants. This includes acoustic echo which can be generated by handsets or speaker phones and line echo which can be generated within the digital (ISDN) to analogue (POTS) conversion within the network. This feature can be set to Off/Normal/High per conference and is set to Normal by default.

- Off – Telephone echo suppression is disabled (same behavior as previous software versions)
- Normal – Telephone echo suppression is enabled
- High – This setting should only be used if telephone echo is still present when set to normal

5.2.6.2 Automatic DTMF Tone Generation

The TANDBERG MPS now supports the automatic generation of DTMF tones to a participant. This is useful when cascading the MPS to an audio or video bridge that is using DTMF passwords for entry access. This feature is available through the Web interface and the API. Allowed digits are 0-9, * and #. Pauses may be inserted by using a comma ",". The pause duration will be 2 seconds for each comma inserted.

5.2.7 Security

5.2.7.1 H.235 Authentication

The TANDBERG MPS now supports H.235 Annex D gatekeeper authentication. This allows the MPS to authenticate itself towards a gatekeeper by supplying a user name and password. New configuration options required for authentication include NTP IP, Authentication ID, and Authentication Password.

5.2.7.2 Caller ID Verification

The TANDBERG MPS now has the ability to restrict calls based on caller ID. By adding caller ID numbers into an active conference, the MPS will only accept calls from endpoints whose caller ID matches the added numbers. This feature requires the Caller ID option on the ISDN PRI circuits if it is to be used in H.320 calls. This feature is also supported on H.323 by using the calling parties E.164 Alias.

5.2.8 Management

5.2.8.1 Resource Management

Included in the J3 release are 2 new API commands for resource management. By default this feature is disabled. This feature allows the MPS to reserve a percentage of its conference and site resources to be used in a scheduling environment. This is intended to prevent Ad-Hoc conferences from consuming all resources when the MPS is used in both Ad-Hoc and Scheduled environments. To enable this feature the following API commands are used.
xConfiguration SystemUnit ResourceManagement: <On/Off>
xConfiguration SystemUnit ResourceManagementPercent: <0-100>

5.3 Changes and Improvements since previous version

5.3.1 Conferencing

5.3.1.1 Additional Conference Templates

The TANDBERG MPS now supports 10 conference templates. Each template can be customized to allow easier creation of different types of conferences. Upon conference creation, select the template you wish to use and all settings will update to reflect the chosen template settings. Furthermore these templates can also be assigned to the Personal Conference entries as well.

5.3.1.2 Conference H323 ID

It is now possible to configure each MPS conference with an H323 ID as well as an E164 Alias. This allows for easier dialing into conferences as well as URI dialing.

5.3.2 Gateway

5.3.2.1 New V35 Configuration Options

With the addition of the MPS Gateway there are now more configuration choices within V35. It is now possible to configure each port with the following:

- Call Control – Changes the RS366 behavior to include Adtran dialing suffixes or to dial entered digits only.
- DTR Pulse – Sets the DTR signal high to allow incoming calls from an Imux or to low to only accept calls when port is added to an active conference.
- Bandwidth – Sets the maximum bandwidth to be used for the port.
- Restrict – Sets port to use 64k or 56k call increments.

5.3.3 Video

5.3.3.1 RTP Time Stamping

The TANDBERG MPS now supports RTP time stamping on H.323 for improved lip sync.

5.3.3.2 H.264 Configuration

It is now possible to disable H.264 at the MPS level. This feature will remove the H.264 video capability for all conferences. This option is available only through the API using the following command.
xConfiguration MCU H264: <On/Off>

5.3.3.3 Voice Switching

Resolved an issue where voice switching would stop when a site held the floor.

5.3.3.4 Conference End Time Icon

Resolved an issue where unexpected restarts could occur when the MPS would display the conference end time icon indicating that the conference is nearing the end time.

5.3.4 Inter-Op

5.3.4.1 QVGA Reporting

Resolved issue where the MPS would report QVGA resolution as Off even though video was being received.

5.3.4.2 4CIF Video

Resolved issue where the MPS would transmit 4CIF video if a Polycom PVX was part of the conference when the transmit video should have been CIF or SIF.

5.3.5 Network

5.3.5.1 RS366 Adtran Suffixing

The TANDBERG MPS now supports the automatic addition of Adtran RS366 suffixes. Now a user does not need to remember complicated suffixes and is able to merely enter the number to dial along with the rate in which to call and the MPS will automatically add the correct dialing suffix for the call rate specified.

5.3.5.2 Dial In by IP Address

It is now possible to configure MPS behavior when receiving an H.323 call that was placed via IP Address. You may configure the MPS to send the call to the Single Number Dial In Ad-Hoc interface or you may direct the call to one of the static conferences. To send the call to Single Number Dial In just add the MPS IP Address into the IP field within the Single Number Dial In configuration. To direct the call to a static conference, the following API command is used.
xConfiguration SystemUnit DefaultIpConference: <0-40>

5.3.5.3 Network Error Handling

The TANDBERG MPS now supports more robust network error handling. This feature is user definable and can be adjusted to suit each environment. This feature allows the possibility to set the MPS into IPLR mode or to block FUR (Fast Update Requests). These features are useful when using in an H.323 environment that is susceptible to packet loss.

5.3.5.4 DNS

J3 now supports up to 5 DNS server IP addresses and 1 DNS Domain Name per IP network interface. These settings are required to facilitate SIP calling.

5.3.6 Audio

5.3.6.1 Audio Input Level

The MPS now reports the audio input level for each site in a call. This is useful in determining sites with low audio input. This level can be seen through the Web interface directly or through the API.

5.3.7 Security

5.3.7.1 Dial Out Password

The MPS now supports conference password protection when using dial out as well as dial in. To enable this feature just set Password Out to On when defining the conference and all dial out sites will then be prompted for the entered password.

5.3.8 Management

5.3.8.1 Improved Web Interface

The Web interface has been improved for easier manageability. It is also now possible to show only active or created conferences on the Overview page as well as search for an active conference by name.

5.3.8.2 Flexible Resource Allocation

Some users do not require all of the features provided by the MPS. It is now possible to increase the MPS capacity by disabling certain features that may not be needed. Below is what will be gained for each feature that is disabled per Media Card.

Features	Number of Conferences per Media Blade	Number of sites per Media Blade
All features	3	16 @ 384kbps
No H.239/Duo Video	3	20 @ 384kbps
No 2nd rate and No H.239/Duo Video	5	20 @ 384kbps
No H.320, No AES/DES	3	16 @ 768kbps

By combining these settings, you can achieve the best performance from each column above. For example, if the user does not require H.239, H.320 (IP only) or H.235 encryption the MPS is capable of providing 3 conferences and 20 sites up to 768kbps per Media Blade.

5.3.8.3 Modular AVO

It is now possible to order AVO per Media Blade. This allows a user to configure a MPS with a fractional resource count for Continuous Presence and H.239/DuoVideo. Please note that configuring a MPS with lower AVO resources than total ports can affect daily use. If a user tries to start H.239/DuoVideo and there are no resources available, the feature will not operate. Please note that when all AVO options are in use all new conferences must be created with Voice Switching, Video Custom Formats set to Off and Dual Video Stream set to Off.

5.3.8.4 Network Module and Network ID Menu

The MPS Web interface now has a Network Module and Network ID drop down menu on the Add Participants dial page. This allows for an easier selection when dialing using these fields. Please note that the Network ID field is used in H.323 only when using dual IP networks and the Network Module and Network ID fields are used with V.35 networks.

5.3.8.5 New Option Key System

J3 has implemented a new option key system. All option keys are now additive so future option upgrades do not require completely new keys. This allows for greater flexibility such as modular AVO. Please note that new option keys will be required when upgrading from a previous MPS software version.

5.4 Supplemental Notes to Manuals

5.4.1 References/Related Documents

TANDBERG Website –http://www.tandberg.net

See the following documents for more information on the TANDBERG MPS:

D1337304 TANDBERG MPS User's Manual
D1333703 TANDBERG MPS800 Installation Sheet
D1369501 TANDBERG MPS200 Installation Sheet
D1363903 TANDBERG MPS API
D1368602 TANDBERG MPS ISDN Interface Kit Installation Sheet
D1368702 TANDBERG MPS Media Port Kit Installation Sheet
D1397401 TANDBERG MPS V35 Ports Kit Installation Sheet
D1390301 TANDBERG MPS Gateway Kit Installation Sheet

5.4.2 Layer 4 Ports used by the system

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 – 6555	TCP	⇔
Video	Range 2326 – 2837	UDP	⇔
Audio	Range 2326 – 2837	UDP	⇔
Data / FECC	Range 2326 – 2837	UDP	⇔
SSH	22	TCP*	⇔
Telnet	23	TCP*	⇔
Telnet Challenge	57	TCP	⇔
HTTP	80	TCP*	⇔
HTTPS	443	TCP	⇔
SNMP (Queries)	161	UDP	⇔
SNMP (Traps)	962	TCP	=> (outgoing from MCU)

(*) Listening sockets

Outgoing H.323 call:
First call uses 5555 for outgoing Q.931 and 5556 for H.245, next uses 5557 for Q.931 and 5558 for H.245, etc.
Incoming H.323 call:
First call uses 5555 for H.245, second 5556 etc. Disconnecting a site in a meeting will not free up available 55XX ports until the whole conference is down.

5.5 Interoperability

The following systems have been tested and verified compatible with this software release

5.5.1 MCU Interoperability

Equipment	Software Revision
Accord MGC 100	7.02.6
Ezenia! MCS 2000	6.3.0.1
TANDBERG MCU	D3.5, D3.6, D3.7, D3.8
RADVision MCU-30	4.0.31
RADVision OnLan MCU	2.2.1.0
Cisco IPVC 3540	4.2.10

5.5.2 Gateway Interoperability

Equipment	Software Revision
TANDBERG Gateway	G2.1, G3.0
RADVision GW-P20	4.0.0.43

5.5.3 Gatekeeper Interoperability

Equipment	Software Revision
CISCO MCM	12.3(10)
TANDBERG Gatekeeper	N3.0, N3.1, N3.2, N4.0
TANDBERG Border Controller	Q2.0, Q2.1, Q2.2, Q3.0
RADVision ECS 100/400	3.5.2.5

5.5.4 Endpoint Interoperability

Equipment	Software Revision
TANDBERG 500-6000/8000	B1.1, B2.3, B3.4, B4.3, B5.1, B5.11, B6.1, B7.0, B8.4, B9.1, B9.2, B10.0, B10.1, E1.1, E2.0, E3.4, E4.1, E4.2, E5.0, E5.1
TANDBERG MXP	F1.5, F2.5, F3.0, F3.1, F3.2, F4.0
Polycom iPower 9000	6.2.0.1208
Polycom iPower 680	6.2.0.1208
Polycom ViewStation FX	6.0.5
Polycom VS	6.0.5
Polycom MP 512	7.5.4
Polycom SP 384	7.5.4
Polycom PVX	8.0.0.0522
Polycom VSX7000	8.0.3
Polycom ViewStation EX	6.0.5
Sony PCS-1	3.14
Sony TL-50	02.10
Aethra Vegastar Silver	6.0.18

5.6 Known Limitations

- Moving a participant that is either a cascade or sending Duo Video may result in conference inconsistencies.
- It is not possible to have more than 100 sites in 1 conference while Conference Selfview is disabled.
- An unexpected restart may occur if a site dials into the Single Number Dial In while the MPS is running at maximum capacity.
- The Polycom MP and SP are unable to call into the Single Number Dial in.
- H.239 does not currently function between the MPS and the Polycom MGC on H.323.
- A Polycom VSX is unable to send H.239 on H.320 calls. It is able to receive H.239 from the MPS. This will be resolved in a future VSX software release.
- An Aethra Vega Star is unable to send H.239 on H.320 calls. It is able to receive H.239 from the MPS. This is currently resolved in the newest Aethra software release.
- Calls fail to connect when using a Rad ECS in full routed mode with H.239 enabled on the MPS.
- The Rad ECS and OnLan gatekeepers do not support H323 prefixes, only E164 prefixes.
- The Cisco MCM does not support H323 prefixes, only E164 prefixes.
- H.239 does not currently function to Sony PCS-1 and TL-50 endpoints on H.320. This is currently resolved in the newest Sony software release.
- When using the Personal Conference prefixes or the H.323 DID feature, the MPS will register to the gatekeeper as a gateway. Please ensure that the prefixes entered do not interfere with your current dialing plan.