TANDBERG MPS 800

Media Processing System Chassis

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TANDBERG

MPS 200
MPS 800

Administrator Guide

Software version J4.5
D13373.10
April 2009

- Join up to 40/160 video and 32/48 audio sites in one or more conferences
- 19” rack-mountable chassis with LCD in front and CompactPCI backplane
- Wide range of network and protocol support: SIP, IP, ISDN and V.35
- Supports H.264 with Continuous Presence and Voice Switching
- Modular and expandable with multiple media processing boards and network interface boards
- Bandwidth: from 56 kbps–2 Mbps
- Supports Simultaneous display of presenter and presentations, Dual Video Stream (DuoVideo™, H.239 or BFCP) including PC presentations using VGA, SVGA and XGA resolutions
- Simple to configure, Plug-and-Play technology
- Supports network and video equipment from multiple vendors
- Outbound, Inbound, and Caller ID password protection
- Supports widescreen HD resolution (1280x720p)
- Flexible design as MCU, Gateway or hybrid
- Highest level of standards based embedded encryption
- Supports TANDBERG Expressway™ firewall traversal, H.460.18 and H.460.19.

What's in this guide?

The top menu bar and the entries in the Table of Contents are all hyperlinks, just click on them to go to the topic.

We recommend you visit the TANDBERG web site regularly for an updated version of this guide.
Go to: http://www.tandberg.com/docs

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Chapter 1

Introduction

Thank you for choosing TANDBERG!

The TANDBERG MPS enables sites on IP, ISDN and High Speed Serial to participate in meetings with each other and offers superior quality and ease of use in one fully-featured multipoint control unit.

How to read this document

You will find that some places information has been copied from other chapters (but adapted, when needed) to let you have all the relevant information there and then. This helps eliminating the need to read through long sections before you can even think of getting started.

Our main objective with this user guide is to address your goals and needs. Please let us know how well we succeeded!

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In this chapter...

What's new in version J4.5?

Version J4 of the TANDBERG MPS provides several new capabilities and enhancements. For your convenience a list of them is provided here.

Software release note

See the MPS Software Release Note for detailed description of the new API commands and features. The software release note is found at the TANDBERG web site.

Go to: http://www.tandberg.com/support/documentation.php?p=Upgrades_and_Diagnostics

Knowledge base

Troubleshooting and FAQs are available at the TANDBERG web site:

Go to: http://www.tandberg.com/support/knowledgebase

MPS API guide

See the MPS API guide for a complete overview of the MPS API commands. The TANDBERG MPS API Guide is found at the TANDBERG web site.

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Security

- Strict password (new API command)
- Legacy encryption mode (new API command)
- Security level (new API command)
- Deny root access (new API command)
- Security log (new API command)
- API login failure lock
- Login banner

Security log files

The MPS now supports security logging. To enable/disable Security logging run the following command:

```
xConfiguration SecurityLog Mode: <On/Off>
```

When enabled the events will be logged to a security log file. The security log file can be accessed from:

- The MPS web interface at `http://x.x.x.x/securitylog` or `https://x.x.x.x/securitylog`,
- or via Telnet, SSH or Serial Port at /var/log/messages while logged in as root.

Older log files are stored in the `/tandberg/log/save/` directory, with filename `message.X`, where `X` is a number.

H264

- Improved support for H.264-only-participants (for example TANDBERG Movi)

IPv6

- HTTP or HTTPS access over IPv6

Stability

- Several bug fixes, see the MPS Software Release Note for detailed descriptions.

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Patent information

The products described in this manual are covered by one or more of the following patents:

US6,584,077	US5,838,664	US5,600,646
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Safety Instructions

For your protection please read these safety instructions completely before you connect the equipment to the power source. Carefully observe all warnings, precautions and instructions both on the apparatus and in these operating instructions.

Retain this manual for future reference.

Water and Moisture

- Do not operate the apparatus under or near water – for example near a bathtub, kitchen sink, or laundry tub, in a wet basement, near a swimming pool or in other areas with high humidity.
- Never install jacks for communication cables in wet locations unless the jack is specifically designed for wet locations.
- Do not touch the product with wet hands.

Cleaning

- Unplug the apparatus from communication lines, mains power-outlet or any power source before cleaning or polishing.
- Do not use liquid cleaners or aerosol cleaners.
- Use a lint-free cloth lightly moistened with water for cleaning the exterior of the apparatus.

Ventilation

- Do not block any of the ventilation openings of the apparatus.
- Never cover the slots and openings with a cloth or other material.
- Never install the apparatus near heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
- Do not place the product in direct sunlight or close to a surface directly heated by the sun.

Lightning

Never use this apparatus, or connect/disconnect communication cables or power cables during lightning storms.

Dust

Do not operate the apparatus in areas with high concentration of dust.

Vibration

Do not operate the apparatus in areas with vibration or place it on an unstable surface.

Power Connection and Hazardous Voltage

- The product may have hazardous voltage inside. Never attempt to open this product, or any peripherals connected to the product, where this action requires a tool.
- This product should always be powered from an earthed power outlet.
- Never connect attached power supply cord to other products.
- In case any parts of the product has visual damage never attempt to connect mains power, or any other power source, before consulting service personnel
- The plug connecting the power cord to the product/power supply serves as the main disconnect device for this equipment. The power cord must always be easily accessible.
- Route the power cord so as to avoid it being walked on or pinched by items placed upon or against it. Pay particular attention to the plugs, receptacles and the point where the cord exits from the apparatus.
- Do not tug the power cord
- If the provided plug does not fit into your outlet, consult an electrician.
- Never install cables, or any peripherals, without first unplugging the device from it's power source.

Servicing

Do not attempt to service the apparatus yourself as opening or removing covers may expose you to dangerous voltages or other hazards, and will void the warranty. Refer all servicing to qualified service personnel.

Unplug the apparatus from its power source and refer servicing to qualified personnel under the following conditions:

- If the power cord or plug is damaged or frayed
- If liquid has been spilled into the apparatus
- If objects have fallen into the apparatus
- If the apparatus has been exposed to rain or moisture
- If the apparatus has been subjected to excessive shock by being dropped
- If the cabinet has been damaged
- If the apparatus seems to be overheated
- If the apparatus emits smoke or abnormal odor
- If the apparatus fails to operate in accordance with the operating instructions.

Accessories

Use only accessories specified by the manufacturer, or sold with the apparatus.

Communication Lines

- Never touch uninstalled communication wires or terminals unless the telephone line has been disconnected at the network interface.
- Do not use communication equipment to report a gas leak in the vicinity of the leak.
- To reduce the risk of fire, use only No. 26 AWG or larger telecommunication line cord (ISDN cables).

Product Approvals

Information about product approvals and CE declarations are found in the Product Approvals in the Appendices section.

Environmental Issues

Thank you for buying a product which contributes to a reduction in pollution, and thereby helps save the environment. Our products reduce the need for travel and transport and thereby reduce pollution. Our products have either none or few consumable parts (chemicals, toner, gas, paper). Our products are low energy consuming products.

TANDBERG's Environmental Policy

Environmental stewardship is important to TANDBERG's culture. As a global company with strong corporate values, TANDBERG is committed to following international environmental legislation and designing technologies that help companies, individuals and communities creatively address environmental challenges.

TANDBERG's environmental objectives are to:

- Develop products that reduce energy consumption, CO_2 emissions, and traffic congestion
- Provide products and services that improve quality of life for our customers
- Produce products that can be recycled or disposed of safely at the end of product life
- Comply with all relevant environmental legislation.

European Environmental Directives

As a manufacturer of electrical and electronic equipment TANDBERG is responsible for compliance with the requirements in the European Directives 2002/96/EC (WEEE) and 2002/95/EC (RoHS).

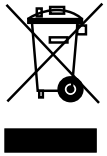
The primary aim of the WEEE Directive and RoHS Directive is to reduce the impact of disposal of electrical and electronic equipment at end-of-life. The WEEE Directive aims to reduce the amount of WEEE sent for disposal to landfill or incineration by requiring producers to arrange for collection and recycling. The RoHS Directive bans the use of certain heavy metals and brominated flame retardants to reduce the environmental impact of WEEE which is landfilled or incinerated.

TANDBERG has implemented necessary process changes to comply with the European RoHS Directive (2002/95/EC) and the European WEEE Directive (2002/96/EC).

Waste Handling

In order to avoid the dissemination of hazardous substances in our environment and to diminish the pressure on natural resources, we encourage you to use the appropriate take-back systems in your area. Those systems will reuse or recycle most of the materials of your end of life equipment in a sound way.

TANDBERG products put on the market after August 2005 are marked with a crossed-out wheelie bin symbol that invites you to use those take-back systems.



Please contact your local supplier, the regional waste administration or http://www.tandberg.com/recycling if you need more information on the collection and recycling system in your area.

Information for Recyclers

As part of compliance with the European WEEE Directive, TANDBERG provides recycling information on request for all types of new equipment put on the market in Europe after August 13th 2005.

Please contact TANDBERG and provide the following details for the product for which you would like to receive recycling information:

- Model number of TANDBERG product
- Your company's name
- Contact name
- Address
- Telephone number
- E-mail.

Digital User Guides

TANDBERG is pleased to announce that we have replaced the printed versions of our user guides with a digital CD version. Instead of a range of different user guides, there is now one CD – which can be used with all TANDBERG products – in a variety of languages. The environmental benefits of this are significant. The CDs are recyclable and the savings on paper are huge. A simple web-based search feature helps you directly access the information you need. In addition, the TANDBERG video systems now have an intuitive on-screen help function, which provides a range of useful features and tips. The contents of the CD can still be printed locally, whenever needed.

China RoHS Table

产品中有毒有害物质表

部件名称	有毒有害物质或元素					
	铅	汞	镉	六价铬	多溴联苯	多溴二苯醚
金属部件	X	O	O	O	O	O
印刷电路板及组件	X	O	O	O	O	O
线缆和线缆组装	X	O	O	O	O	O
显示器（包括照明灯）	X	X	O	O	O	O

说明:

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除非在产品上有另外特别的标注，以下标志为针对所涉及产品的环保使用期限标志。环保使用期限只适用于产品在产品手册中所规定的使用条件。

Features overview and MPS capacity

This Administrator Guide is provided to help you make the best use of your TANDBERG MPS, Media Processing System.

The TANDBERG MPS enables sites on IP (H.323 and SIP), ISDN and High Speed Serial (V.35/ RS449/RS530 w/RS366 support) to participate in meetings with each other, and at the same time it offers superior quality and ease of use in one fully-featured multipoint control unit, MCU.

The TANDBERG MPS may also include the optional Gateway functionality.

MPS Models

The TANDBERG MPS can be found in two models, the MPS 800 a 9U-sized unit, and the MPS 200 a 3U-sized unit.

The two models differ in the size of the chassis and the number of boards that they can host. However there are no differences in the feature set. Therefore in this manual, we will use the term TANDBERG MPS to refer to both models, unless a specific situation requires referring to each model with its own name.

Main Features

- IP, ISDN PRI, Leased E1/T1 (G.703) and High Speed Serial (V.35/RS449/RS530 w/RS366 support) networks are supported at call rates of up to 2 Mbps for each call.
- Up to 40 simultaneous conferences with the MPS 800, and up to 10 simultaneous conferences with the MPS 200.
- Up to 160 video sites and 48 telephony calls with the MPS 800 and up to 40 video sites and 32 telephony calls with the MPS 200 can be supported at the same time in some configurations, each benefiting from the same superb audio and video quality. The TANDBERG MPS can also be used purely as an audio-bridge.
- Secure Conference[TF] - using standard based AES 128 and DES encryption. Support both H.235 v2 and v3 in the same conference.
- Best Impression[TF] - Automatic selection of layout and resolution depending on number of meeting participants.
- Numerous different conference layouts, 16:9 wide formats and Voice Switched mode.
- Dual Video Stream - support for DuoVideo[TF], H.239 and BFCP.
- DuoVideo[TF]/H.239/BFCP - automatically distributed to conference participants supporting these protocols. Support for mix of DuoVideo[TF] and the H.239 or BFCP protocols in same conference. Endpoints not supporting these protocols will receive main stream.
- Downspeeding[TF] - if channels are dropped during a videoconferencing session, the connection is automatically re-established without interruption.
- Audio and video transcoding to the best quality available.
- Secure Access - support SSH, XML/SOAP over HTTPS, Web (HTTP) encrypted password. The Telnet, SSH, HTTP, HTTPS and SNMP services can be disabled.
- Web-interface for system management, call management, diagnostics, multi language and software uploads.
- Worldwide compatibility with standards-based videoconferencing systems.
- Gateway functionality – Embedded gateway with up to 80 Gateway calls on the MPS 800 and up to 20 Gateway calls on the MPS 200.
- Ad Hoc functionality and Single number dial in, with waiting room and dynamic access and authorization mechanisms. Possibility to pre-configure up to 500 personal conference and service prefix for dynamic allocation of personal conference.
- Up to double bandwidth capacity on IP only, non encrypted calls.
- Support for participant identification in video, with localizations support (Chinese, Traditional Chinese, Thai, Japanese, Korean and Russian).
- Encoding support for High Definition Continous Precence resolution.
- Optimal Voice Switch - Video switching, providing point-to-point quality.

Options

Simplifies scheduling and the use of video meeting resources through highly automated functionality:

- Management using TANDBERG Management Suite.
- Scheduling using TANDBERG Scheduler, Microsoft® Outlook®, Microsoft® Office Communicator® or IBM Lotus Notes®
- Ad Hoc conferencing through Microsoft® Office Communicator®

TANDBERG MPS Capacity

The TANDBERG MPS 800 can support up to

- 40 simultaneous conferences
- 160 simultaneous video calls
- 48 simultaneous telephone calls
- 80 simultaneous Gateway calls

The TANDBERG MPS 200 can support up to

- 10 simultaneous conferences
- 40 simultaneous video calls
- 32 simultaneous telephone calls
- 20 simultaneous Gateway calls

TANDBERG MPS Capacity on IP

The maximum bandwidth on IP for each Media Processing Board is 15360 kbps. With 8 Media Processing Boards installed in a MPS 800 the maximum bandwidth on IP is 122800 kbps.

Setting Encryption to On will decrease the maximum bandwidth throughput, but not the total number of ports. The maximum bandwidth for each of the Media Processing Boards is 7680 kbps with Encryption set to On in all calls.

NOTE: Encrypted SIP calls is a Beta feature, and you should not run with Encrypted SIP calls in a production environment, it has limited support and is for evaluation and testing use only.

TANDBERG MPS Capacity on ISDN

The maximum bandwidth for ISDN for each Media Processing Board is 7680 kbps. With 4 E1/ T1 ISDN Interface Card installed in a MPS 800, and 8 Media Processing Boards the maximum bandwidth is 61440 kbps.

One V.35 Serial Interface Card could handle maximum 61440 kbps in maximum 32 calls. More V.35 Serial Interface Card would increase, not the bandwidth capacity, but the number of possible calls up to a maximum of 128 calls. The Gateway capacity is 7680 kbps per Media.

In a secure conference, there is no support for telephone participants.

To increase the capacity, the MCU can be connected in a cascade. See Appendices > Appendices > Distributed MCUs for details.

TF - TANDBERG First

TANDBERG MPS at a Glance

Rack Mountable Chassis

The TANDBERG MPS chassis is 19" rack-mountable.

- On the front of the chassis is a Liquid Crystal Display (LCD) for initial configuration and basic system information.
- There are four Light Emitting Diodes (LEDs) indicating the power status.
- The backplane of the chassis is provided with advanced CompactPCI technology for high speed communication between the boards.
- There are three cooling fans in the lower front of the chassis.
- The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis that can host up to 8 Media Processing Boards and up to 4 Network Interface Cards.
- The TANDBERG MPS 200 has 3U-19" rack-mountable chassis that can host up to 2 Media Processing Boards and up to 2 Network Interface Cards.

System Controller Board - Front view

The System Controller Board is installed in the first slot in the chassis.

- In the MPS 800 the first slot is the first from the left of the chassis.
- In the MPS 200 the first slot is the first from the bottom of the chassis.

It is very important that the System Controller Board is installed in the first slot in the chassis! Installing in any other slot can damage the System Controller Board.

The System Controller Board takes care of the following functions:

- Call control
- System management
- The embedded Web server

The System Controller Board is equipped with the following interfaces:

- 1 X LAN / Ethernet (RJ-45) 10/100 Mbit on the front.
- 2 X LAN / Ethernet (RJ-45) 10/100 Mbit on the back (only 1 in use, Enet2)
- 1 x COM port on the front
- 2 X USB port (these are for future use)

The LAN interface on the System Controller Board is for management/call control signalling. Note that management is disabled on Enet2. This interface is only for call control. The 2 LAN interfaces will allow you to connect to two non-overlapping IP-networks so that participants with no IP-routing between them can be joined in the same conference. At least one Media Processing Board must be connected to each network. The 2xLAN interfaces will give the TANDBERG MPS support for two Gatekeepers, one on each network. To use the COM1 port you need a RJ-45 to RS-232 converter.

Media Processing Board

Add-on boards for media processing are installed in adjacent slots in the chassis. The Media Processing Boards handles the following functions:

- Video processing, see Video Features in the Appendices section for details.
- Audio processing, see Audio > Create Conference in the Using the MPS section for details.
- Transcoding, see Transcoding and Ratematching in the Appendices section for details.
- Encryption, see Secure Conference (Encryption) in the Appendices section for details.
- Continuous Presence/Voice Switching, see Video Features in the Appendices section for details.

Each of the Media Processing Boards is equipped with 1xLAN interface for H.323 and SIP media. You will also find 4 Light Emitting Diodes (LEDs) for board status. With the TANDBERG MPS 800, there is support for up to 8 Media Processing Boards. With the TANDBERG MPS 200, there is support for up to 2 Media Processing Boards. See the Appendices section for further details on the Media Processing Board.

Power Units

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units for configurations of 1 to 3 Media Processing Boards. If the unit has more than 3 Media Processing Boards the TANDBERG MPS 800 has to be equipped with 3 hot-swappable power units. The power units are installed at the back of the chassis. You will also find the power switch/connector at the back of the chassis.

The TANDBERG MPS 200 is always shipped with 1 power unit integrated in the chassis.

System Controller Board - Rear View

The second LAN interface of the System Controller Board is accessible from the rear side.

See the Appendices section for further details on the System Controller Board.

Network Interface Cards

The Network Interface Cards of the TANDBERG MPS are installed from the rear panel.

There are two types of Network Interface Cards:

- PRI E1/T1 ISDN Interface Card (IIC-8).
  Each PRI E1/T1 ISDN Interface Card has 8 x PRI interfaces.
- V.35 Serial Interface Card (SIC-32).
  Each of the V.35 Serial Interface Card has 32 x V.35/RS366 ports.

There is support for up to 4 Network Interface Cards.

There can be a mix of PRI E1/T1 ISDN Interface Cards and V.35 Serial Interface Cards.

The PRI E1/T1 ISDN Interface Card and the V.35 Serial Interface Card may only be installed in slot 1-6. (1 being the first Media Processing Board, left from the System Controller Card, seen from the rear).

See the Appendices section for further details of the different Network Interface Cards.

Chapter 2

Getting started

This section describes the first time installation of the MPS - Media Processing System. The section covers unpacking, what's in the box, mounting the MPS, cable connections, starting up the system and initial IP configuration using LCD. You will also find pictures of the MPS 800 and MPS 200 with a description of the interfaces.

The Quick Setup will help you get your MPS online and operational quickly. It provides a step-by-step guide to the basic H.323 services setup via the MPS web interface.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

In this chapter...

Precautions, Unpacking and Mounting

Precautions

- Never install communication equipment during a lightning storm.
- Never install jacks for communication cables in wet locations unless the jack is specifically designed for wet locations.
- Never touch un-installed communication wires or terminals unless the communication line has been disconnected at the network interface.
- Use caution when installing or modifying communication lines.
- Avoid using communication equipment (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lightning.
- Do not use communication equipment to report a gas leak in the vicinity of the leak.
- The socket outlet shall be installed near the equipment and shall be easily accessible.
- Never install cables without first switching the power OFF.
- This product complies with directives: LVD 73/23/EC and EMC 89/366/EEC.
- Caution - Double pole fusing.
- Power must be switched off before power supplies can be removed from or installed into the unit.

Unpacking

To avoid damage to the unit during transportation, the TANDBERG MPS is delivered in a special shipping box.

The shipping box contains the following components:

1. Chassis:
   a. MPS 200, a 3U chassis with power supply.
   b. or MPS 800, a 9U chassis with 2 or 3 x Power Units (depending on the number of Media Processing Boards installed)
   c. 1 x System Controller Board
   d. The Media Processing Boards ordered
   e. PRI E1/T1 ISDN Interface Card (if ordered)
   f. V.35 Serial Interface Card (if ordered)
2. Administrator Guide and other documentation on CD
3. Installation sheets
4. 4 screws and 4 nuts for rack mounting and 4 pads
5. Cables:
   a. Power cable
   b. Ethernet cables
   c. ISDN cables (optional)
   d. V.35 kit (optional). The kit includes cables that convert from high-density connectors on V.35 card to TANDBERG's standard V.35 connectors (26pin DSUB) and 19" rack-mountable panel where the V.35 connectors will fit.
   e. RJ45 to RS-232 converter cable

Preparations on Site

The mounting space must be prepared before you start:

- Make sure the TANDBERG MPS is accessible and that all cables can be easily connected
- For ventilation: Leave a space of at least 10cm (4 inches) behind the TANDBERG MPS's rear panel and 10cm (4 inches) in front of the front panel
- The room in which you install the TANDBERG MPS should have an ambient temperature between 0°C and 35°C (32°F and 95°F) and between 10% and 90% non-condensing relative humidity
- Do not place heavy objects directly on top of the TANDBERG MPS
- Do not place hot objects directly on top, or directly beneath the TANDBERG MPS
- Use a grounded AC power outlet for the TANDBERG MPS.

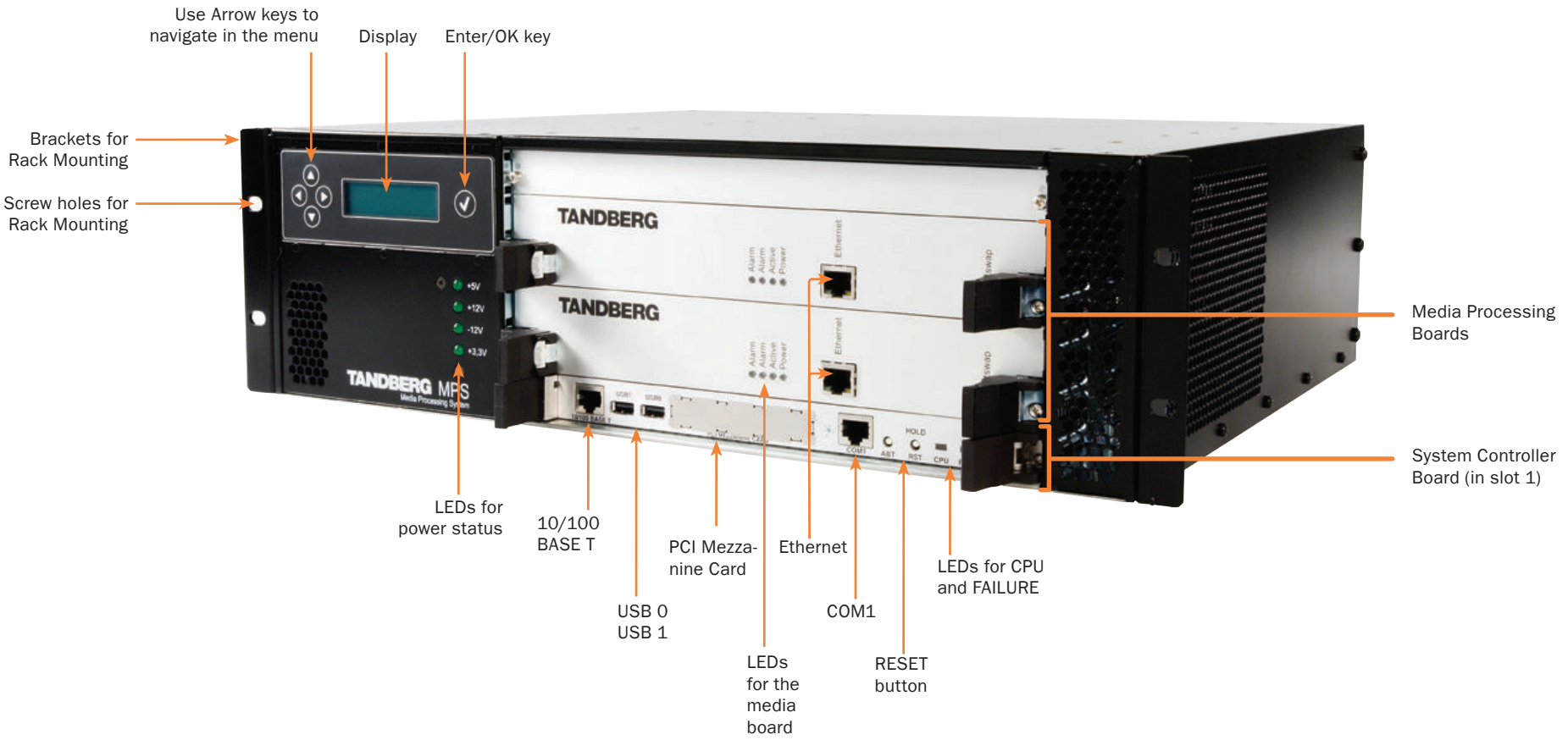
Mounting the MPS on a Rack

The TANDBERG MPS comes with 4 screws and 4 nuts for mounting in standard 19" racks. The chassis is equipped with brackets.

1. Before starting the rack mounting, please make sure the TANDBERG MPS is placed securely on a hard, flat surface.
2. Disconnect the AC power cable.
3. Make sure that the mounting space is according to the Preparations on Site in the section above.
4. Insert the chassis into a 19" rack, and secure with screws in the front (four screws) and nuts (four nuts).

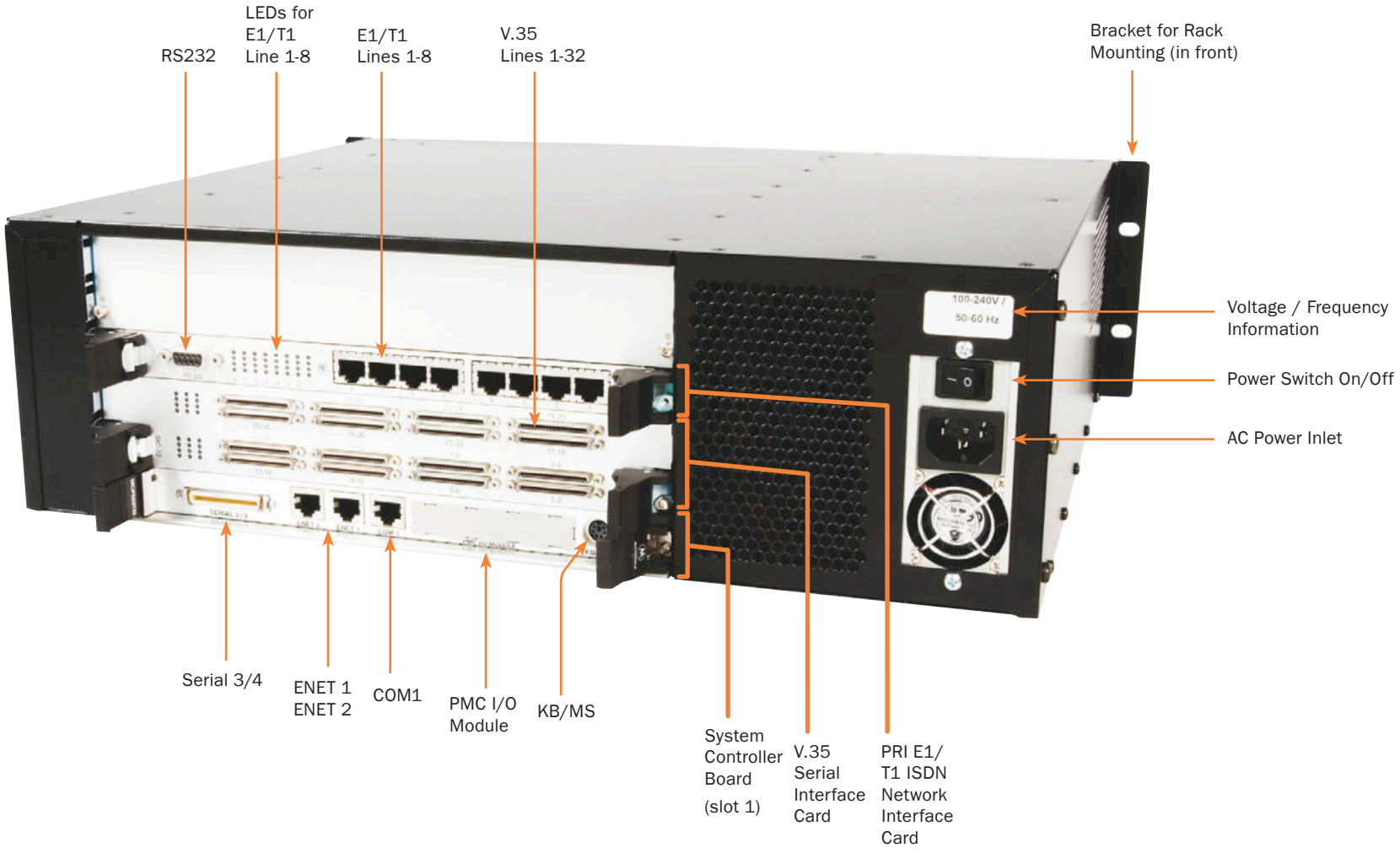
System overview

MPS 200 with 2 Media Processing Boards



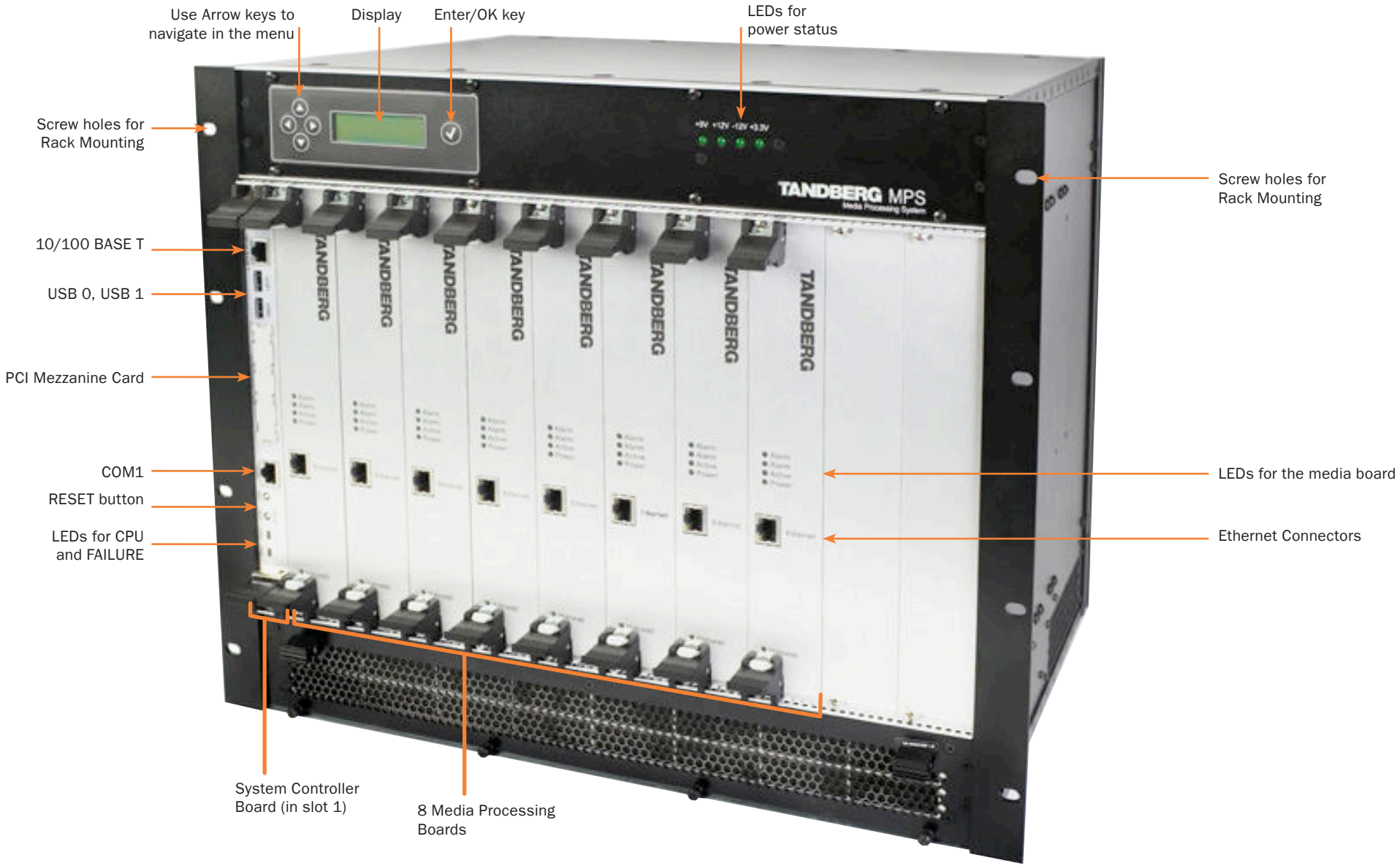
System overview, *cont...*

MPS 200 with ISDN and V.35 Network Cards



System overview, *cont...*

MPS 800 with 8 Media Processing Cards



System overview, *cont...*

MPS 800 with ISDN and V.35 Network Cards

Connecting cables

Power Cable

Connect the system power cable to an electrical distribution socket.

LAN Cables

Connect the LAN cable from the 10/100 BASE T connector on the front of the System Controller Board to your network.

From the Ethernet connector on the front of each of the Media Processing Boards, connect a LAN cable to your network.

NOTE: Use a switch/router and not a hub for connecting LAN cables between the TANDBERG MPS and the rest of your network

Connect to Two Separate IP Networks

If you want to connect the TANDBERG MPS to two separate IP networks you must use the second 'Enet2' interface on the back of the System Controller Board, in addition to the 'Enet1' interface on the front side.

ISDN-PRI Cables

For each of the PRI interfaces, the E1/T1 cable should be connected to a CSU (Channel Service Unit). You will need a CSU between the TANDBERG MPS and the PRI line from your network provider.

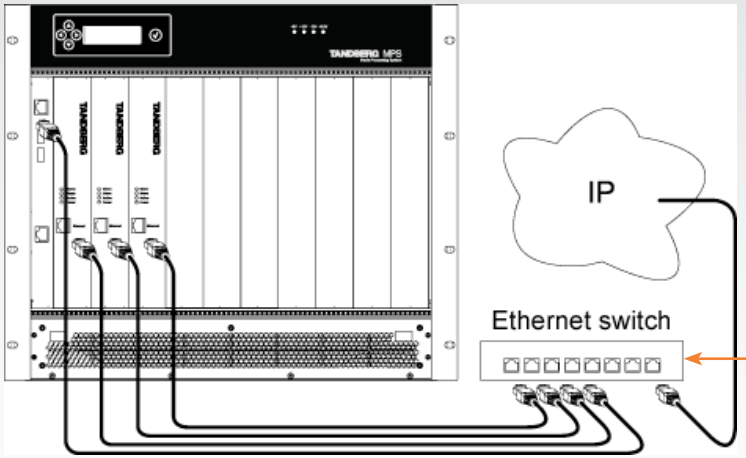
NOTE: Both Leased E1/T1 (G.703) and ISDN PRI uses the same physical interface on the ISDN Interface Card.

V.35 Cables

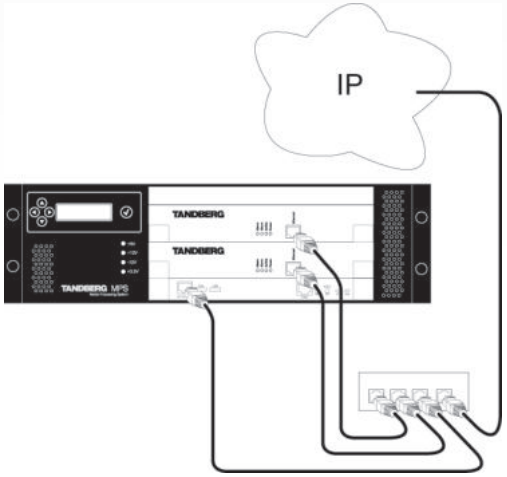
Connect the high-density connector on the V.35 card and insert the TANDBERG standard V.35 connectors (26pin DSUB) into the corresponding position in the 19" rack-mountable panel. Additional cables will be required for connection to customer provided device.

See the Appendices section for further details on the V.35 cables.

Connecting Ethernet Cables



Do not use a hub!
Use a switch/router and not a hub for connecting LAN cables between the TANDBERG MPS and the rest of your network.



Starting up the system

Preparations

Before starting the TANDBERG MPS make sure the following has been done:

1. The System Controller Board must be installed in the first slot in the chassis.
2. Make sure the Media Processing Board(s) and the Network Interface Card(s) are interted correctly into the chassis.
2. MPS 800: Make sure the Power Unit(s) are inserted correctly into the back of the chassis.
3. Connect the Power Cable.

Turning on the MPS

Set the Power Switch on the back of the chassis to 1.

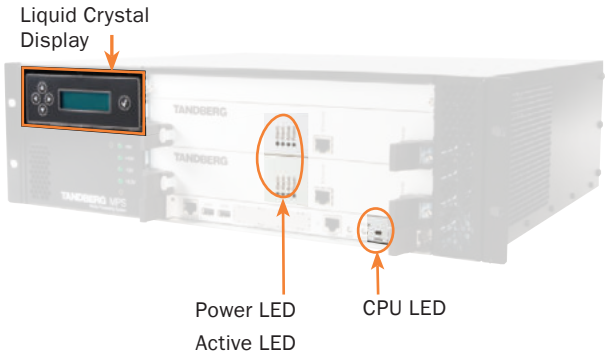
Startup mode

- On the front of the chassis you will see the CPU LED on the System Controller Board flashing
- On the Media Processing Boards the Power LED turns green.
- The Active LED is flashing green while software is being uploaded to the Media Processing Boards.
- In the LCD (Liquid Crystal Display) you will see the TANDBERG logo.

The system is ready for configuration

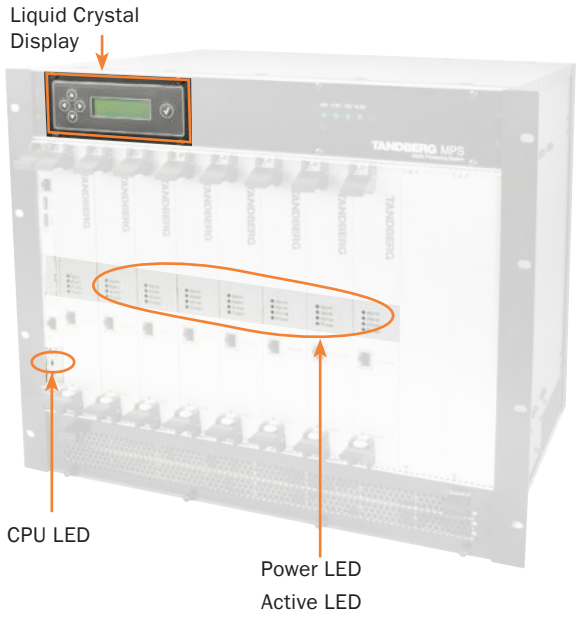
1. When both Power and Active LED's on the Media Processing Boards are steady green
2. And the initial configuration menu appears on the LCD.

MPS200 Front Side

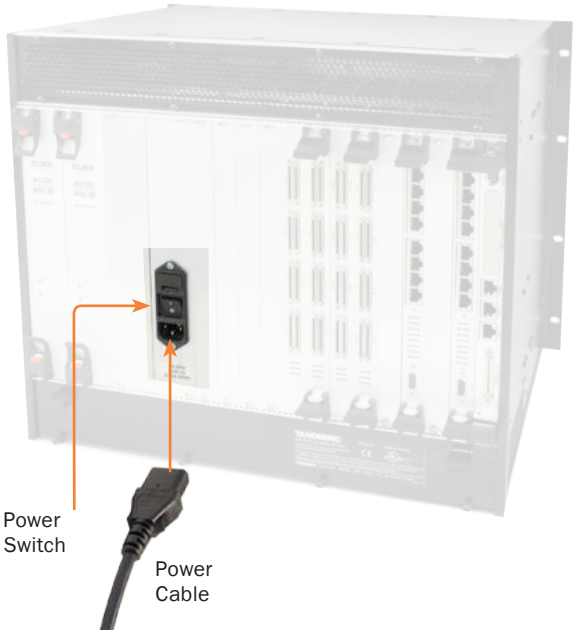


MPS200 Back Side

MPS800 Front Side



MPS800 Back Side



The LCD interface

LCD - Liquid Crystal Display

The initial configuration is done through the Liquid Crystal Display (LCD) on the front of the chassis.

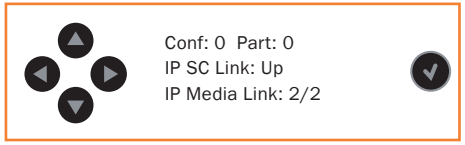
Status Page

Conf - number of active conferences

Part - number of active participants

IP SC Link - system controller ethernet link indicator

IP Media Link - media card ethernet link indicator



Example with a MPS having 2 Media boards

Parameter Configuration

The initial configuration of the MPS needs to be done through the LCD. The initial configuration is done in the Parameter Config menus.

Parameter Info

The configuration can be viewed from the Parameter Info menu.

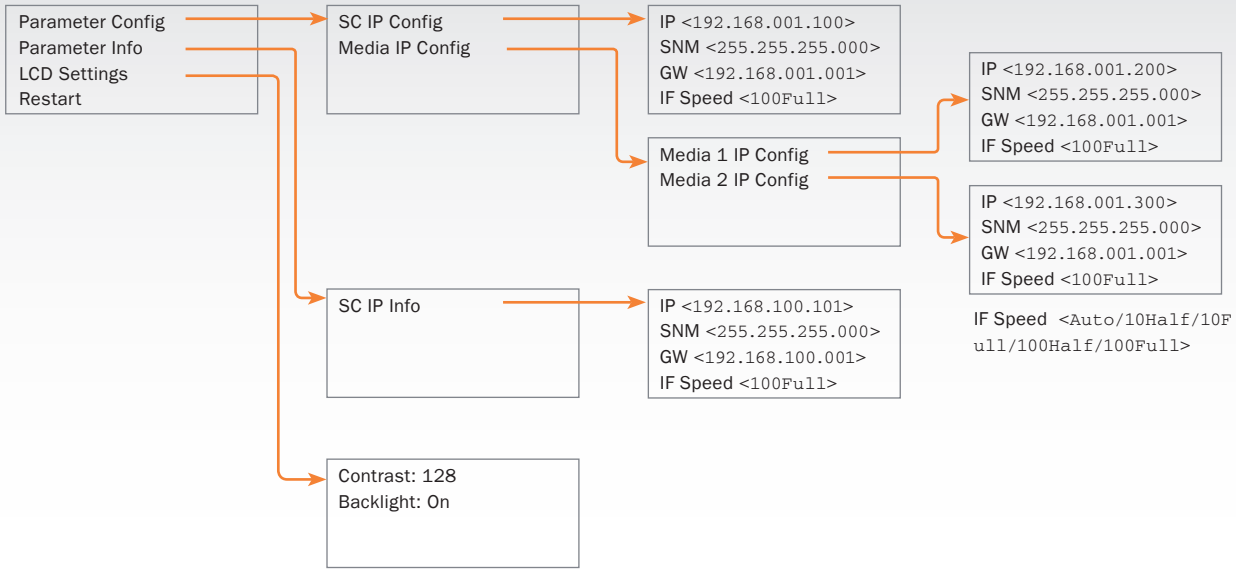
LCD Settings

The LCD can be configured to control the Contrast and Backlight.

Restart

Select Restart to restart the MPS.

Example of the LCD menu structure on a MPS having 1 System Controller Board and 2 Media Processing Boards:



LCD buttons

- Press Arrow Up to move up in the menu or to increase a digit (0 –> 9)
- Press Arrow Down to move down in the menu or to decrease a digit (9 –> 0)
- Press Arrow Left to step back in the menu or to move the next digit to the left
- Press Arrow Right to step forward in the menu or to move the next digit to the right
- Press Confirm button to confirm a selection

Take care! When changing a value it is automatically saved.

If Contrast is set to a very low or a very high value the display will become black.

Restart To activate changes to the Parameter Config the MPS requires a restart.

System Controller Board

Initial Configuration

LCD

Before using the TANDBERG MPS it requires some basic configuration. This is done through the Liquid Crystal Display (LCD) on the front of the chassis.

Static Network Addresses

Follow the instructions, to the right, to configure static network addresses and Ethernet speed of the System Controller Board (SC) and Media Processing Board.

Access to the MPS

After configuring the IP-address of the System Controller Board (and restart the MPS), you may continue the configuration from the web interface. You have access to the TANDBERG MPS by entering the IP-address of the System Controller Board in a standard Web-browser.

Enter Password

You will be asked to enter a password. Some Web-browsers requires a User Name to be entered. Could be such as 'admin'. The default password for the TANDBERG MPS is "TANDBERG". The password is case sensitive.

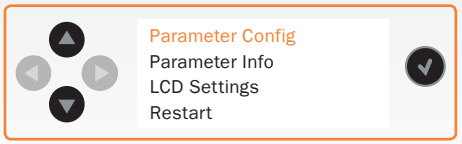
Change Password

To change the password of the system, you will need to log on to the MPS using a command line interface. For further information see the TANDBERG MPS API guide. Go to: http://www.tandberg.com/docs.

Parameter Configuration

1

- Press any key to open the main menu.
- Use Arrow buttons to select from the sub-menu and press the Confirm button.



Parameter Config - Configuration of System Controller Board(s) and Media Processing Board(s).

Parameter Info - See the configuration of the System Controller Board(s) and Media Processing Board(s).

LCD Settings - Configuration of the LCD.

Restart - Restart the MPS

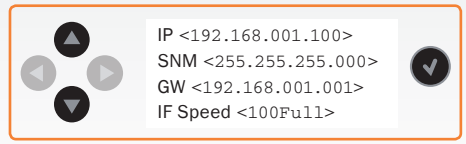
2

Use Arrow buttons to select SC IP Config and press the Confirm button.

SC IP Config - Configuration of the System Controller Board.

3

Use Arrow buttons to select a line and set the three addresses and the Ethernet speed. Use Arrow Left to step back in the menu.



IP: Defines the static IP address of the System Controller Board: Example <192.168.001.100>

SNM: Defines the static IP Subnet Mask for the network: Example <255.255.255.000>

GW: Defines the static IP Gateway address: Example <192.168.001.001>

IF Speed: Defines the Ethernet speed: <Auto/10Half/10Full/100Half/100Full>

LCD buttons

- Press Arrow Up to move up in the menu or to increase a digit (0 –> 9)
- Press Arrow Down to move down in the menu or to decrease a digit (9 –> 0)
- Press Arrow Left to step back in the menu or to move the next digit to the left
- Press Arrow Right to step forward in the menu or to move the next digit to the right
- Press Confirm button to confirm a selection

4

Restart the TANDBERG MPS

To activate changes to the Parameter Configuration the MPS requires a restart.

- Use Arrow Left to step back to the main menu and select Restart

Media Processing Board

Initial Configuration

LCD

Before using the TANDBERG MPS it requires some basic configuration. This is done through the Liquid Crystal Display (LCD) on the front of the chassis.

Static Network Addresses

Follow the instructions, to the right, to configure static network addresses and Ethernet speed of the Media Processing Board.

Access to the MPS

After configuring the IP-address of the System Controller Board (and restart the MPS), you may continue the configuration from the web interface. You have access to the TANDBERG MPS by entering the IP-address of the System Controller Board in a standard Web-browser.

Enter Password

You will be asked to enter a password. Some Web-browsers requires a User Name to be entered. Could be such as 'admin'. The default password for the TANDBERG MPS is "TANDBERG". The password is case sensitive.

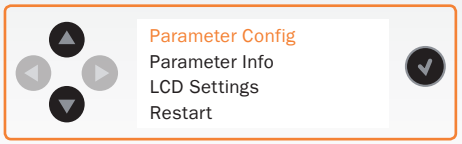
Change Password

To change the password of the system, you will need to log on to the MPS using a command line interface. For further information see the TANDBERG MPS API guide. Go to: http://www.tandberg.com/docs.

Parameter Configuration

1

- Press any key to open the main menu.
- Use Arrow buttons to select from the sub-menu and press the Confirm button.



Parameter Config - Configuration of System Controller Board(s) and Media Processing Board(s).

Parameter Info - See the configuration of the System Controller Board(s) and Media Processing Board(s).

LCD Settings - Configuration of the LCD.

Restart - Restart the MPS

2

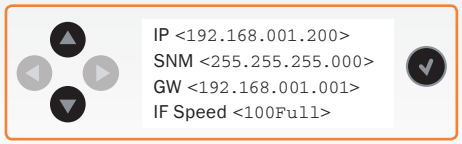
Use Arrow buttons to select Media IP Config and press the Confirm button.

Media IP Config - Configuration of the Media Processing Board (MPS 200 having 1-2 boards and MPS 800 having 1-8 boards).

If more than one Media board the menu will be extended with the number of boards installed. Use Arrow buttons to select the desired board and continue the configuration.

3

Use Arrow buttons to select a line and set the three addresses and the Ethernet speed. Use Arrow Left to step back in the menu.



IP: Defines the static IP address of the Media Processing Board: Example <192.168.001.200>

SNM: Defines the static IP Subnet Mask for the network: Example <255.255.255.000>

GW: Defines the static IP Gateway address: Example <192.168.001.001>

IF Speed: Defines the Ethernet speed: <Auto/10Half/10Full/100Half/100Full>

LCD buttons

- Press Arrow Up to move up in the menu or to increase a digit (0 –> 9)
- Press Arrow Down to move down in the menu or to decrease a digit (9 –> 0)
- Press Arrow Left to step back in the menu or to move the next digit to the left
- Press Arrow Right to step forward in the menu or to move the next digit to the right
- Press Confirm button to confirm a selection

4

Restart the TANDBERG MPS

To activate changes to the Parameter Configuration the MPS requires a restart.

- Use Arrow Left to step back to the main menu and select Restart

The web interface

The TANDBERG MPS is normally controlled via the web interface, but any authorized user may schedule meetings on the TANDBERG MPS by using the TANDBERG Management Suite (TMS) or the TANDBERG Scheduler.

In this guide we will describe how to use the web interface of the TANDBERG MPS.

The TANDBERG MPS consists of the Multipoint Control Unit (MCU), and may also include optional Gateway (GW) functionality.

Note that the Gateway Configuration tab on the web interface is only visible when the Gateway option is installed.

SSH and Telnet

The TANDBERG MPS can also be controlled via SSH or Telnet by using a comprehensive set of API commands. This enables the TANDBERG MPS to be controlled by a different user interface, such as an AMX or Crestron control systems.

For the web interface of the TANDBERG MPS to work properly, make sure you have set the permission to 'Allow Scripting' in the Security Settings in the Web browser.

Read about the API commands in the TANDBERG MPS API Guide. Go to: http://www.tandberg.com/docs.

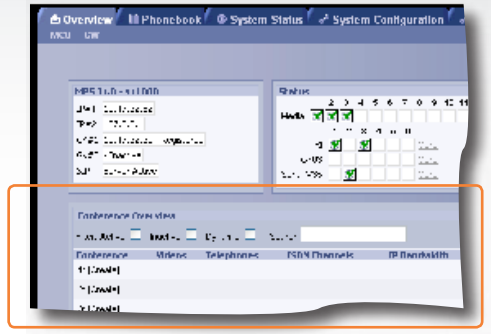
1. Start a Web browser and enter the IP address of the MPS.

   You can use either **http** or **https**, with **https** recommended for administrative access.

   The **http** settings can be configured after the first login. Go to: System Configuration > Miscellaneous.

2. When prompted for login, enter the user name **admin** and password, **TANDBERG** (default password).

3. The Overview page is the opening page and you will see all conferences currently active on the TANDBERG MPS.



Overview

The Overview page lets you see information about all conferences currently active on the TANDBERG MPS.

MCU

- Shows all MCU conferences

GW*

- Shows all Gateway calls.

Phone Book

Use Phone Book to:

- Add new entry
- Edit existing entry
- Add up to 250 single entries
- Add up to 16 group entries.

System Status

Gives an overview of:

- H.320 Status
- PRI Status
- G.703 Status
- V.35 Status
- Media Board Information
- H.323 Status
- SIP Status
- System Information

System Config.

Lets you configure:

- H.320
- PRI
- G.703
- IP
- Media Board
- Serial V.35
- H.323
- SNMP
- SIP
- QoS
- Miscellaneous
- Upgrade
- Language
- XML
- Certificate Mgmt

Gateway Config.*

Lets you configure:

- Dialling Rules
- Settings
- Files

MCU Configuration

Lets you configure:

- Dial In Numbers
- Network Profile
- Conference Template
- Files

* Visible only when a Gateway is installed

Quick setup

Step 1: IP Configuration

Configuration of the MPS using Web Interface

Open a web browser and type in the IP address to access the web interface. The IP address to enter was configured from the LCD during the installation process.

Configuration of the System Controller Board

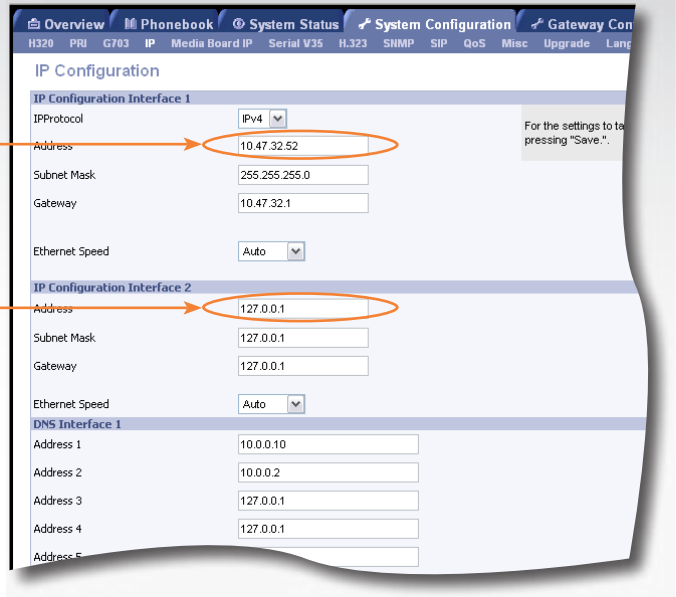
Navigate to System Configuration > IP:

- Verify the IP Address set via the front panel LCD.
- If applicable, set the IP Address of Interface 2.

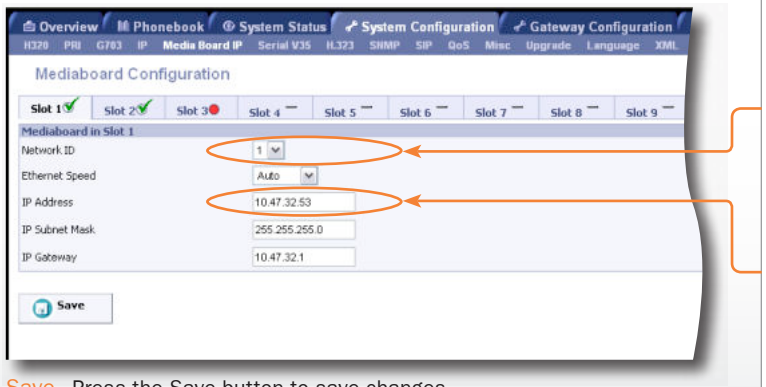
As a minimum you must configure Interface 1. If your TANDBERG MPS is connected to two different IP networks, you must configure both Interface 1 and Interface 2.

For further information, go to: System Configuration > IP Configuration.

1



2



Save - Press the Save button to save changes.

Configuration of the Media Board(s)

Navigate to System Configuration > Media Board IP:

- Make sure to select the correct Network ID.

When using only 1 IP network all Media Boards should be set to Network ID 1.

- Set the IP Address for each of the installed Media Boards.

For further information, go to: System Configuration > Media Board IP.

Quick setup, *cont...*

Step 2: H.323 Configuration

Configuration of the MPS using Web Interface

Open a web browser and type in the IP address to access the web interface. The IP address to enter was configured from the LCD during the installation process.

Registration to a Gatekeeper

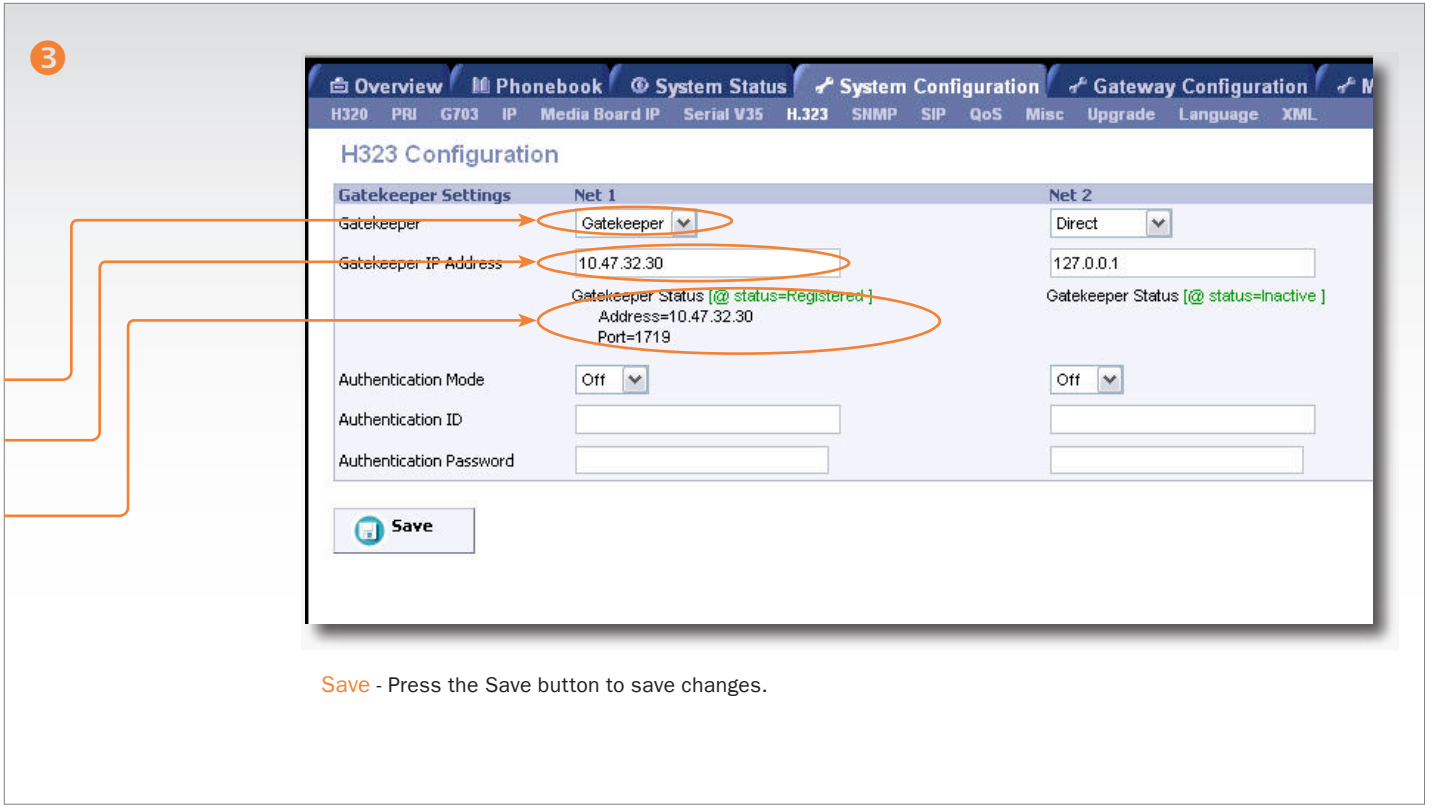
Navigate to System Configuration > H.323:

- Set the Gatekeeper mode to Gatekeeper to enable the system to register to a Gatekeeper.
- Enter the Gatekeeper IP Address.

When registered, the H.323 Gatekeeper Status shows Registered, displays the Gatekeeper's IP address and the port used.

Problems with registration will be shown as Registering and a Red alarm on the Conference Overview page.

For further information, go to: System Configuration > H.323 Configuration.



Save - Press the Save button to save changes.

Quick setup, *cont...*

Step 3: PRI Configuration

Configuration of the MPS using Web Interface

Open a web browser and type in the IP address to access the web interface. The IP address to enter was configured from the LCD during the installation process.

Simple PRI Configuration

Navigate to System Configuration > PRI:

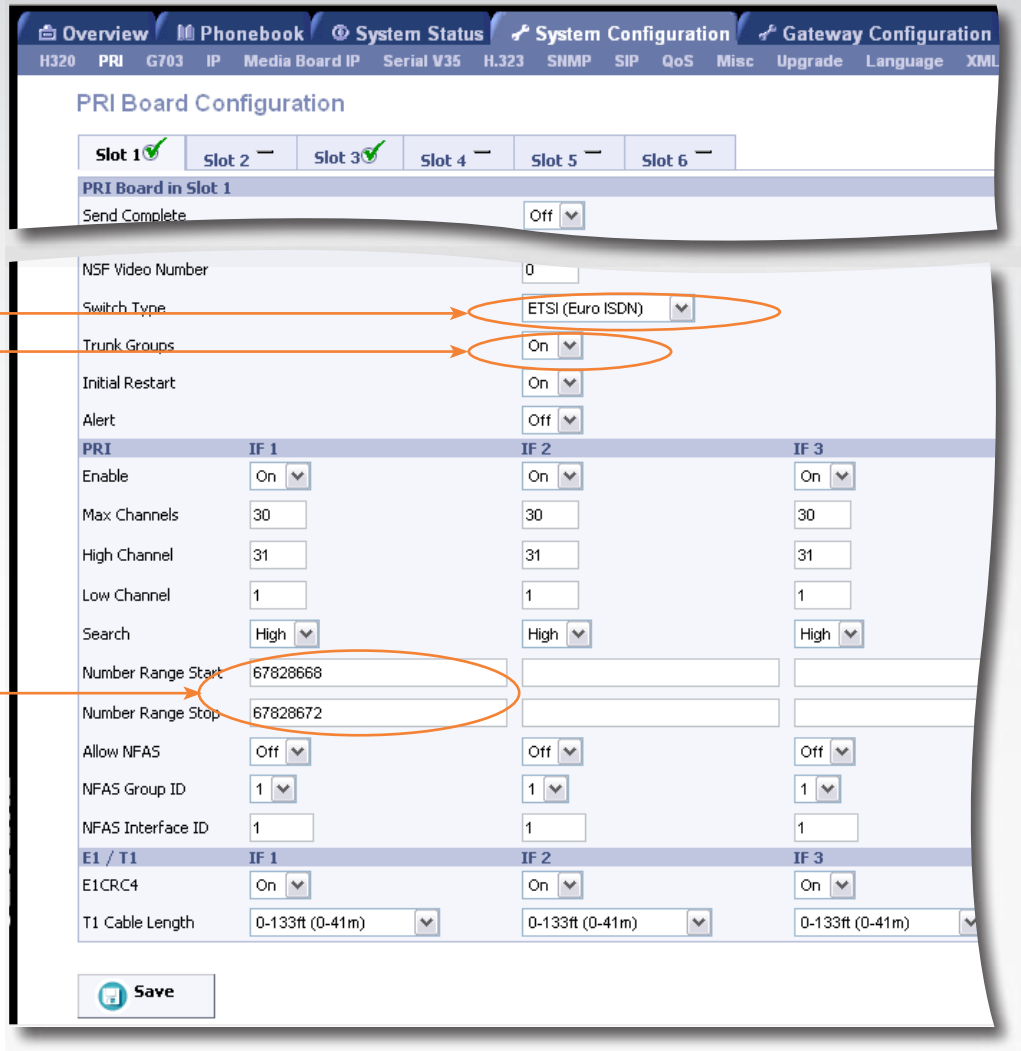
- Set Switch Type by selecting the apropriate PRI protocol: National ISDN, AT&T Custom ISDN, ETSI (Euro ISDN) or Japan/Taiwan ISDN.

All PRI lines on the same E1/T1 Interface Card must use the same PRI Protocol.

- If same number for all PRI's set Trunk Groups to On.
- Configure PRI Numbers in Number Range Start and Number Range Stop.

For further information, go to: System Configuration > PRI Board Configuration.

4



Save - Press the Save button to save changes.

Quick setup, *cont...*

Step 4: Dial In Configuration

Configuration of the MPS using Web Interface

Open a web browser and type in the IP address to access the web interface. The IP address to enter was configured from the LCD during the installation process.

Dial In Configuration

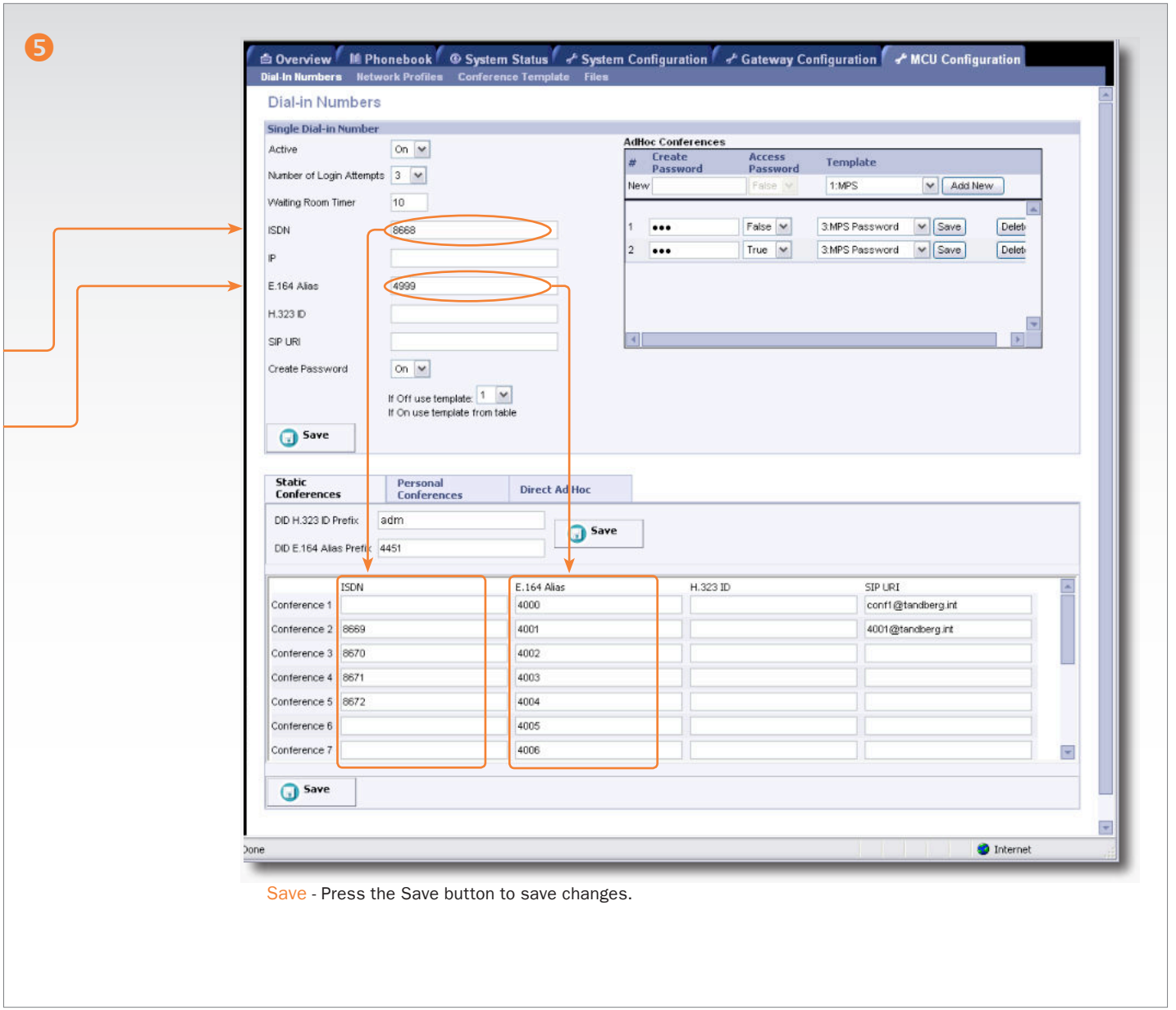
Navigate to MCU Configuration > Dial In Numbers:

Configure one or more of the following numbers and aliases to enable the MPS to know which conference number to dial into.

- ISDN Number. The ISDN number must be a valid PRI number of the TANDBERG MPS at hand.
- E.164 Alias. The H.323 E.164 numeric alias for each conference.

The Single Dial In Number allows the MPS to set up multiple conferences, in an Ad Hoc manner by the user dialling in to a single number.

You may configure and create different types of conferences. For further information, go to: MCU Configuration > Dial In Numbers.



Save - Press the Save button to save changes.

Chapter 3

Using the MPS

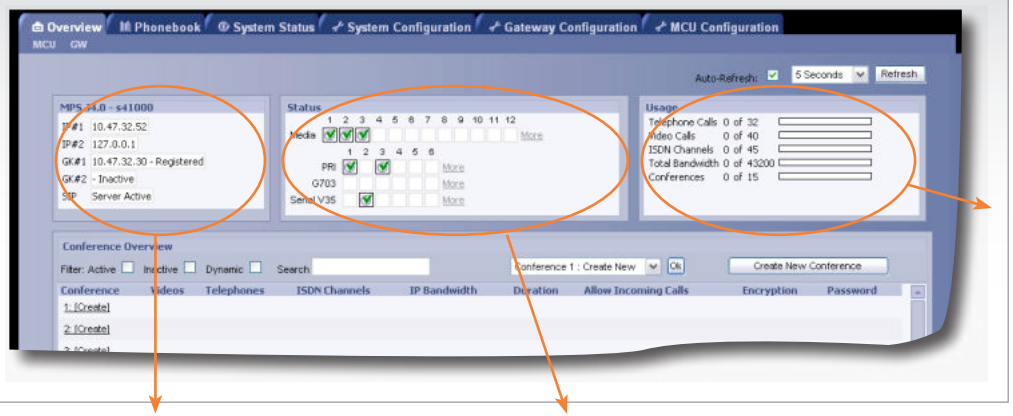
The Using the MPS section will help you understand how to use the MCU, Gateway and Phone Book. It provides a step-by-step guide to the basics as well as a description of each setting available from the web interface.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

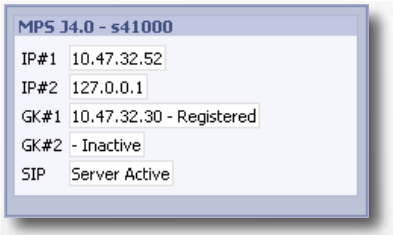
In this chapter...

MCU Overview, Usage Information



The General field displays some basic information:

- Software version
- IP address(es)
- Gatekeeper address and status
- SIP status



The Status field displays the connections status for the installed boards and cards:

- Media boards
- ISDN PRI card
- ISDN G.703 card
- SERIAL card

Indicates OK

Indicates a warning

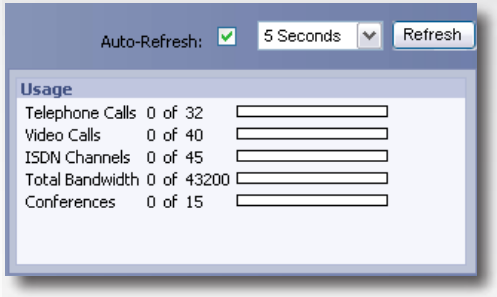
Indicates an error

Click More to see more details.

The Usage pane for MCU calls shows the current status of all the available resources (Video, Telephone, ISDN Channels and Total Bandwidth).

Right above the Usage pane you can:

- Set Auto-Refresh to On/Off
- Adjust the Refresh Rate (2-30 seconds)
- For a quick refresh click the Refresh button.



The status of the MCU resources:

- Telephone Calls 0 of 32: indicate that 0 telephone calls are connected to the MCU. The total number of supported telephone calls in this configuration is 32.
- Video Calls 0 of 40: indicate that 0 video calls are connected to the TANDBERG MPS. The total number of supported video calls in this configuration is maximum 40.
- ISDN Channels 0 of 45: indicate that 0 ISDN channels are used on the TANDBERG MPS. The total number of supported ISDN channels in this configuration is 45.
- Total Bandwidth 0 of 43200: indicate that 0 kbps bandwidth is used on the TANDBERG MPS. The maximum bandwidth available in this configuration is 43200 kbps.
- Conferences 0 of 15: indicate that there are up to 15 available conferences on this TANDBERG MPS, and that 0 of them are currently in use.

MCU Conference Overview

Filter Conference Overview

- Active If selected all the active conferences are shown
- Inactive If selected all the inactive conferences are shown
- Dynamic If selected all the Ad Hoc created conferences are shown

Create a Conference

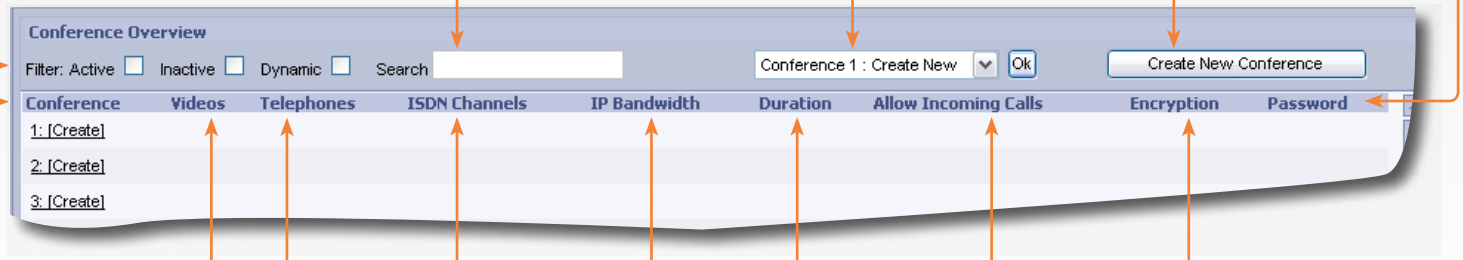
1. Click on the Conference#: Create New drop down list to select a conference and press OK button to create a new conference.
2. Click on the Create New Conference button to create a new conference.
3. Click on [Create] in the conference list to set up a new conference.

Password For privacy, a conference can be password protected.

Indicates that a valid password is required for all participants dialling in to the conference.

Indicates that the conference is not password protected.

Search View a specific conference selected by name using the Name search field.



Conference Status When a conference is active click on the corresponding conference name or number to see conference status in details. It is possible to change the conference name.

Videos Shows the number of connected video sites and the maximum number of allowed video sites in the conference.

Telephones Shows the number of connected telephones and the maximum number of allowed telephones in the conference.

ISDN Channels Shows the total number of ISDN channels used in the conference.

IP Bandwidth Shows the total IP bandwidth used in the conference.

Duration Shows the call duration of the conference.

Allow Incoming Calls

- Yes Participants can dial in to the conference.
- No Incoming calls are not allowed. The MCU must dial out to all participants.

Encryption (Secure Conference[TF]) The MCU supports Secure Conference DES and AES.

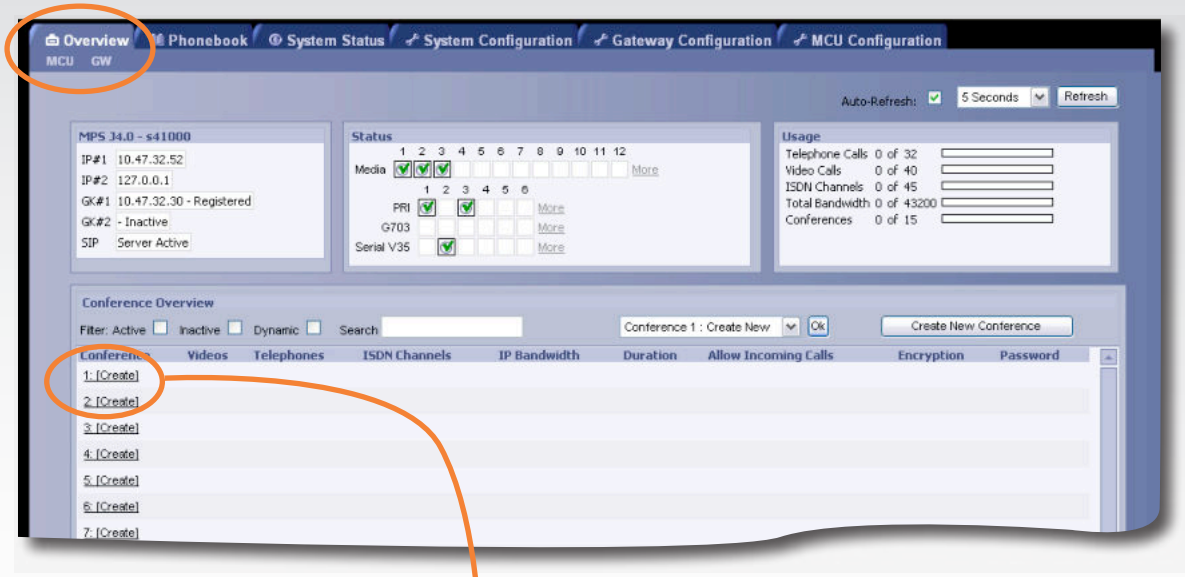
Indicates that DES (56 bit) encryption is used.

Indicates that AES (128 bit) encryption is used.

Indicates that the conference is not encrypted.

Create a conference on the MCU

Select the Overview tab and MCU

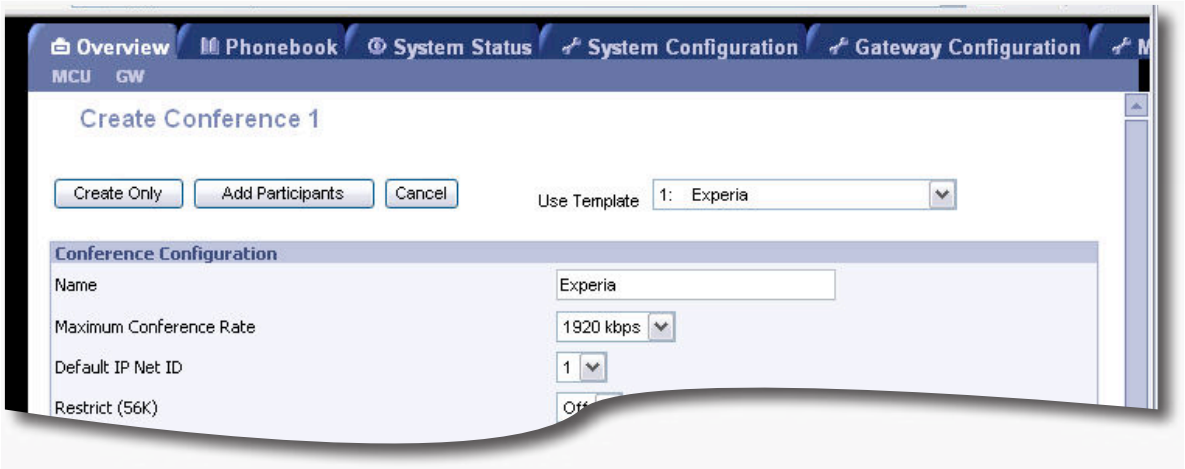


Create a new conference

1. Click on [Create] in the conference list to set up a new conference.
2. or: click on the Create New Conference button to create a new conference.
3. or: click on the Conference#: Create New drop down list to select a conference and press OK button to create a new conference.

Create a Conference

Click on [Create] in the conference list to create a new conference.



Create Conference 1

All settings on the Create Conference # view will be explained in detail on the following pages.

Create conference

Conference Configuration

Name

The conference name will be shown on the Conference Overview page and on the Conference Status page.

Maximum Conference Rate

Custom Selection: Specifiy the maximum possible call rate allowed in the conference. If a participant does not support this rate the MCU will connect at the highest rate possible.
Telephone: When Telephone is selected, an audio bridge will be created and no video participants will be able to join.

Default IP Net ID

Specifies which IP-network ID to use as default.

Restrict (56K)

Non-restricted and restricted calls are supported in the same conference. It is thus possible to select restrict for each call individually when dialling.

On: Set Restrict (56K) to On to make the MCU to set up restricted calls by default.

Off: Set Restrict (56K) to Off to make the MCU to set up non-restricted call and down-speed to 56 kbps if necessary.

Allow Incoming Calls

On: Set to On to allow incoming calls. Incoming call will be automatically answered.

Off: Set to Off to automatically reject all incoming calls.

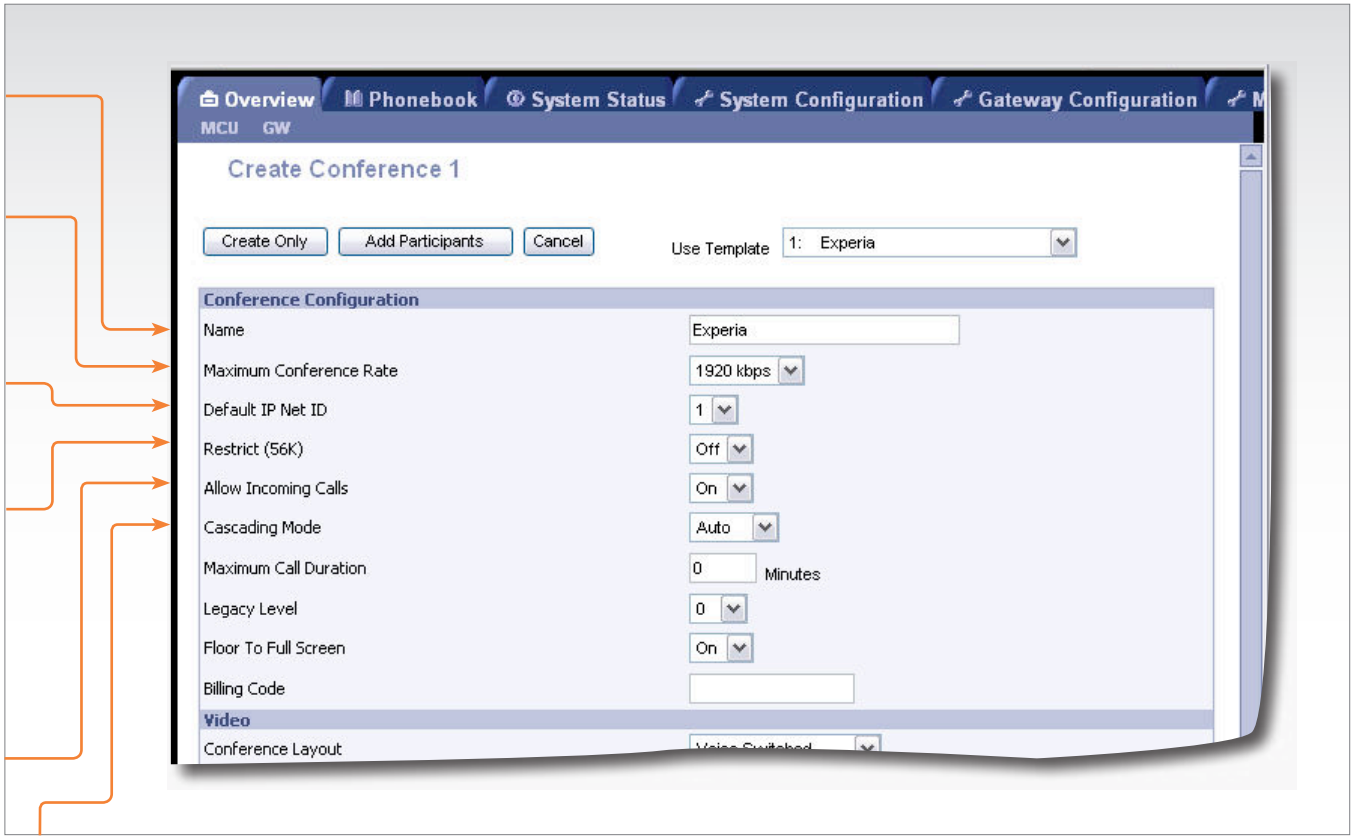
Cascading Mode

Used to join two or more conferences together.

Auto: Set to Auto to automatically determine which conference is 'master' and which conference(s) are 'slave'. The 'master' conference will have control over the video layout. When left in 'Auto' mode, the conference dialling in to the other conferences, will become the 'master'.

Master: Set to Master when this conference is the one controlling the video layout for the whole conference. It is not recommended to have more than one 'master' in a conference.

Slave: Set to Slave when another conference manually has been assigned 'master'. The slave will be forced to Full Screen voice switched mode.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

The conference is then ready to start from the Conference Overview with the comment [Inactive].

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Use Template

Predefined conference configurations can be used by selecting one of the 10 predefined Conference templates in the Create Conference pane.

The Conference Template 1 is default. All settings can be manually edited by an authorized user.

Create conference, *cont...*

Conference Configuration

Maximum Call Duration

Determines the maximum duration (minutes) of the conference.

- All sites will be disconnected when the specified 'Max Call Duration' has been reached.
- 10 minutes, five minutes and one minute prior to this, a warning will be displayed to all the video participants in the conference, indicating the remaining time.
- The conference will remain active, after having timed out, allowing sites to dial in again and restart the conference timer.
- The conference administrator can extend the time.
- The timer for the max call duration will not begin until the first participant is connected.

Legacy Level

When connecting older videoconferencing endpoints to the MCU, problems can occur since older equipment sometimes do not handle modern capabilities.

- It is possible to disable Dual Stream Switching setting the Legacy Level to a value of 1-15.
- When set to 0-7: All capabilities are sent from the MCU
- When set to 8-14: The H.264 capability is disabled.
- When set to 15: The only capabilities sent for level 15 are H.261, G.711 and G.722.

Floor to Full Screen

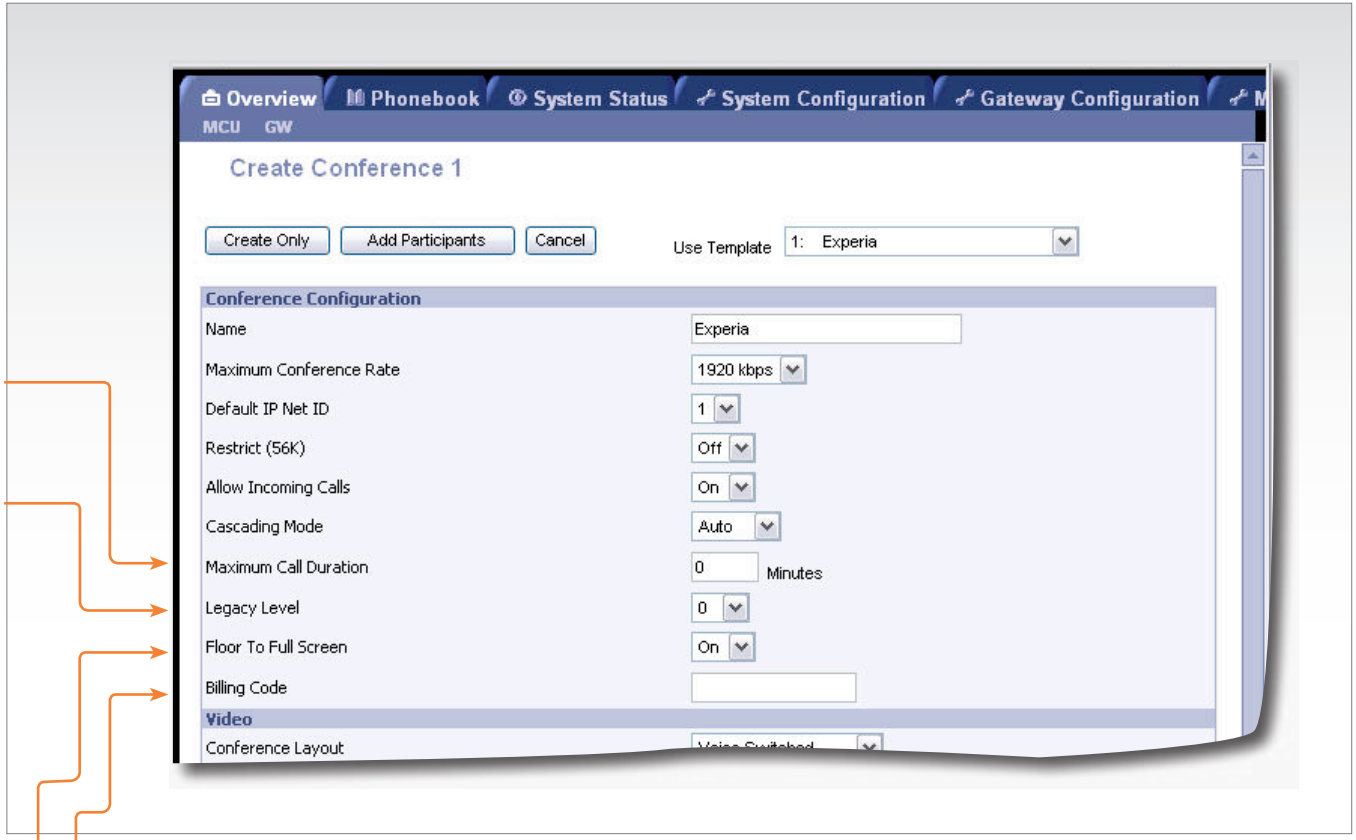
On: When set to On, the participant requesting the floor will be shown in full screen to all the other video participants, regardless of current speaker. The same will happen if the conference administrator assign floor to a site.

Off: When set to Off, the participant requesting the floor will be shown in the largest sub-picture if there is one in the selected layout.

Billing Code

When defining a conference, a specific billing code can be assigned to the conference.

All calls in this conference will be associated with this billing code. This will allow management tools, such as the TANDBERG Management Suite, to use the code for billing purposes.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

The conference is then ready to start from the Conference Overview with the comment [Inactive].

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

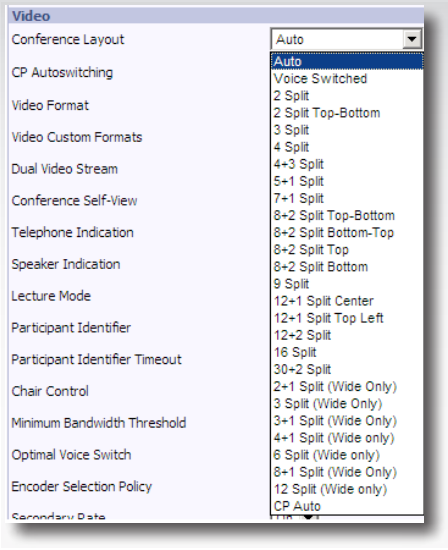
Use Template

Predefined conference configurations can be used by selecting one of the 10 predefined Conference templates in the Create Conference pane.

The Conference Template 1 is default. All settings can be manually edited by an authorized user.

Create conference, *cont...*

Video, Conference Layout



Conference Layout

Auto: When set to Auto the most suitable conference layout will automatically be selected depending on the total number of participants in the actual conference.

Voice Switched: Full Screen voice switched will show the current speaker in full screen to all the other participants, regardless of how many participants there are in the conference. Current speaker will see the previous speaker.

Custom Selection: Select a specific Conference Layout for the conference. The different selections are illustrated to the right.

CP Auto: When set to CP Auto there will be a dynamic change in layout dependent on the number of sites in the conferense. The CP Auto will start with VS->CP4->CP9->CP16.

Show Current Speaker

The screen will be split into a specified number of sub-pictures. The currently speaking participant will be shown in the largest sub-picture in asymmetric layouts. With fewer participants than the total number of sub-pictures, the empty sub-pictures will be black. If there are more participants than the total number of sub-pictures, only the last speakers will be displayed.

Create conference, *cont...*

Video settings

CP Autoswitching

The CP Autoswitching enables you to swap non speaking sites with the least active sites in the picture. This lets you see all participants in a conference, even if they are not speaking.

CP Autoswitching can be set to a value between 0 seconds (default) and 60 seconds. The number of seconds denotes how long each of the remaining participants shall be displayed on the screen.

If set to 0 seconds the CP Autoswitching will be disabled.

Note that the CP Autoswitching will be performed in the least active sub picture if one or more of the participants speak.

Video Format

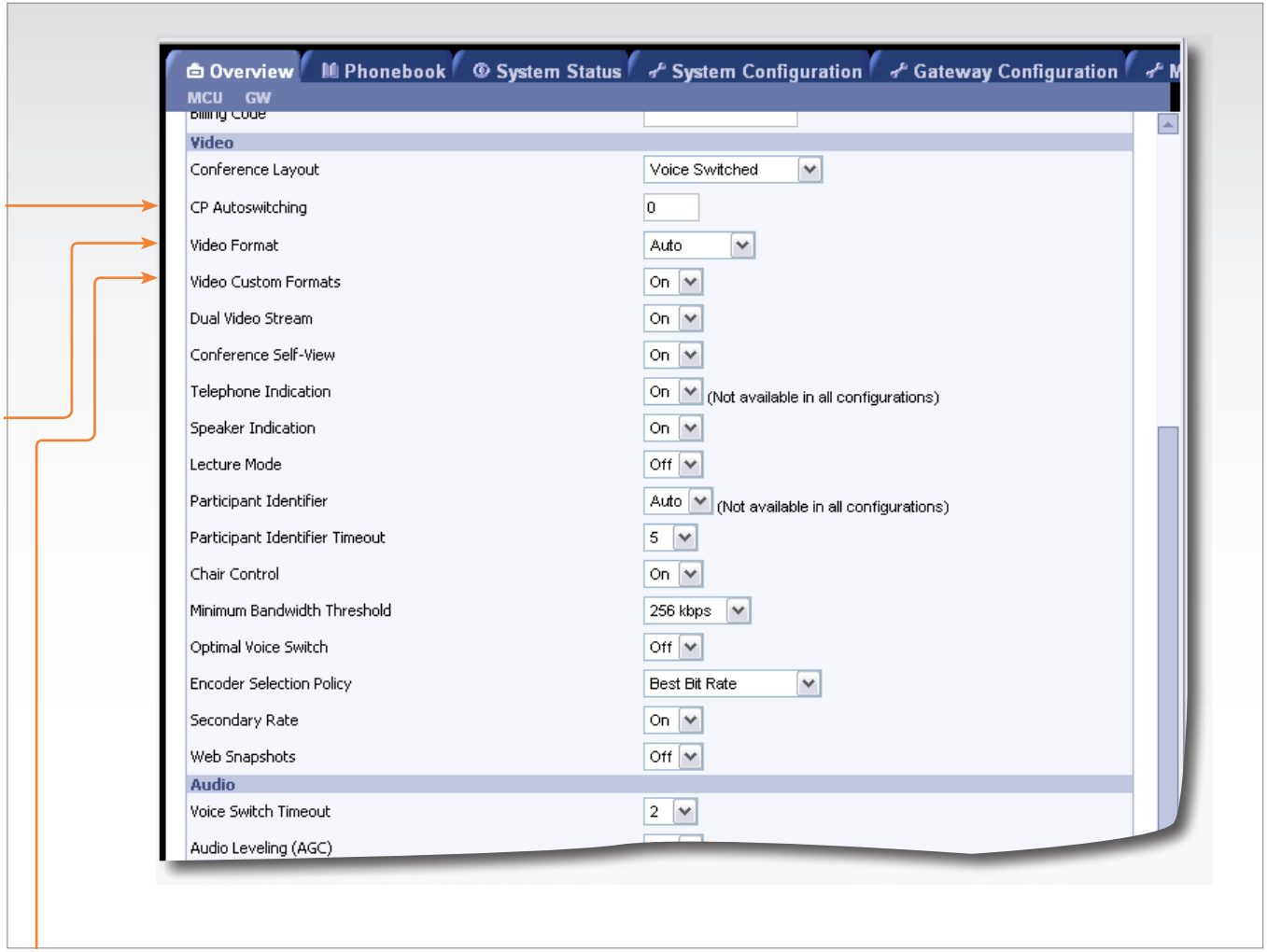
Defines the video format for Continuous Presence (CP) mode.

Auto (Best Impression[TF]) In Continuous Presence mode the MPS will select Motion (CIF) if the call rate is below 256 kbps and sending 4:3 aspect ratio. When sending 16:9 aspect ratio the MPS will select Motion (w288p) if the call rate is below 512 kbps. At call rates of 256 kbps and higher the MPS will select Sharpness (4CIF) when sending 4:3 aspect ratio. When sending 16:9 aspect ratio the MPS will select Sharpness (w576p) at call rates of 512 kbps and higher.

Motion: Set to Motion to prioritize motion and show up to 30 fps in CIF resolution and transmit the highest common format, preferably H.264 CIF when sending 4:3 aspect ratio or H.263+ w288p when sending 16:9 aspect ratio.

Sharpness: Set to Sharpness to prioritize crisp and clear picture and transmit the highest common format, preferably H.263+ 4CIF when sending 4:3 aspect ratio or H.263+ w576p when sending 16:9 aspect ratio.

In Full Screen Voice Switched Conference layout, the MCU will prioritize H.264 CIF as the highest common format.



Video Custom Formats

On: Set to On to support custom formats, such as SIF and VGA resolutions. It allows true resolution to be maintained, rather than being scaled to another format. This is of particular benefit to users of NTSC and VGA resolutions, ensuring that their images are not scaled to fit with the PAL standard.

Off: Set to Off when support for custom formats are not needed.

Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Video settings

Dual Video Stream

The MCU supports DuoVideo[TF], H.239 and BFCP.

On: Set to On to enable a Dual Video Stream protocol for this conference. Both DuoVideo[TF] and H.239 or BFCP are supported in the same conference.

Off: When set to Off, Dual Video Stream will not be supported in this conference.

Conference Selfview

On: Set to On to enable Conference Selfview. The users will see themself in the picture when more than one participant is in the conference.

Off: Set to Off to disable Conference Selfview.

Telephone Indication

On: Set to On to enable a Telephone Indicator to be displayed when there are telephone (audio only) participants connected to the conference. When the telephone participant is speaking the indicator will be outlined.

Off: Set to Off to disable the Telephone Indicator to be displayed.

Speaker Indication

On: Set to On to enable a Speaker Indicator, a coloured line, to be displayed around the sub-picture that will indicate who is the currently speaking participant.

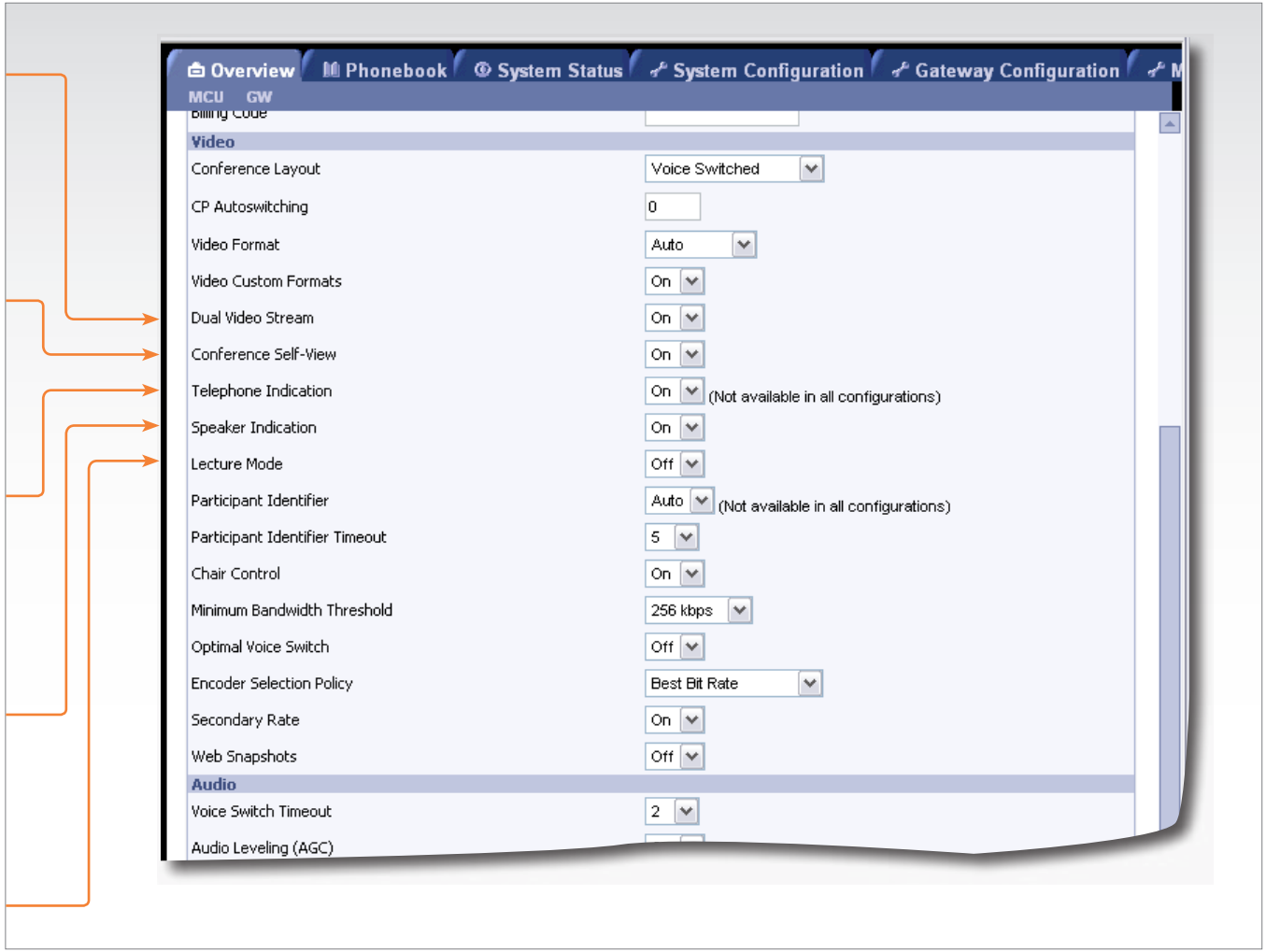
Off: Set to Off to disable the coloured line to be displayed.

Lecture Mode

On: Set to On to enable the Lecturer to be displayed in full screen to the other participants.

- The Lecturer is the participant which is assigned floor.
- The Lecturer will see a scan of all the participants in a full screen view or one of the supported sub-picture views. To enable the scan of other sites the CP Autoswitching must be set.

Off: Set to Off to disable the Lecturer, the participant which is assigned floor, to be view in full screen.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Video settings

Participant Identifier

Auto: Set to Auto to let the System Name of a participant to be displayed the number of seconds set in Participant Identifier Timeout.

On: Set to On to enable the System Name for each participant to be displayed in the picture during the conference.

Off: Set to Off to disable the System Name to be displayed.

Participant Identifier Timeout

Set the number of seconds (1 - 30 seconds) the Participant Identifier will be visible, if set to auto. The identifier will re-appear at every picture changing event.

Chair Control

On: Set to On to enable Chair Control. The conference supports H.243 and BFCP Chair Control functionality initiated from the participants connected to the conference.

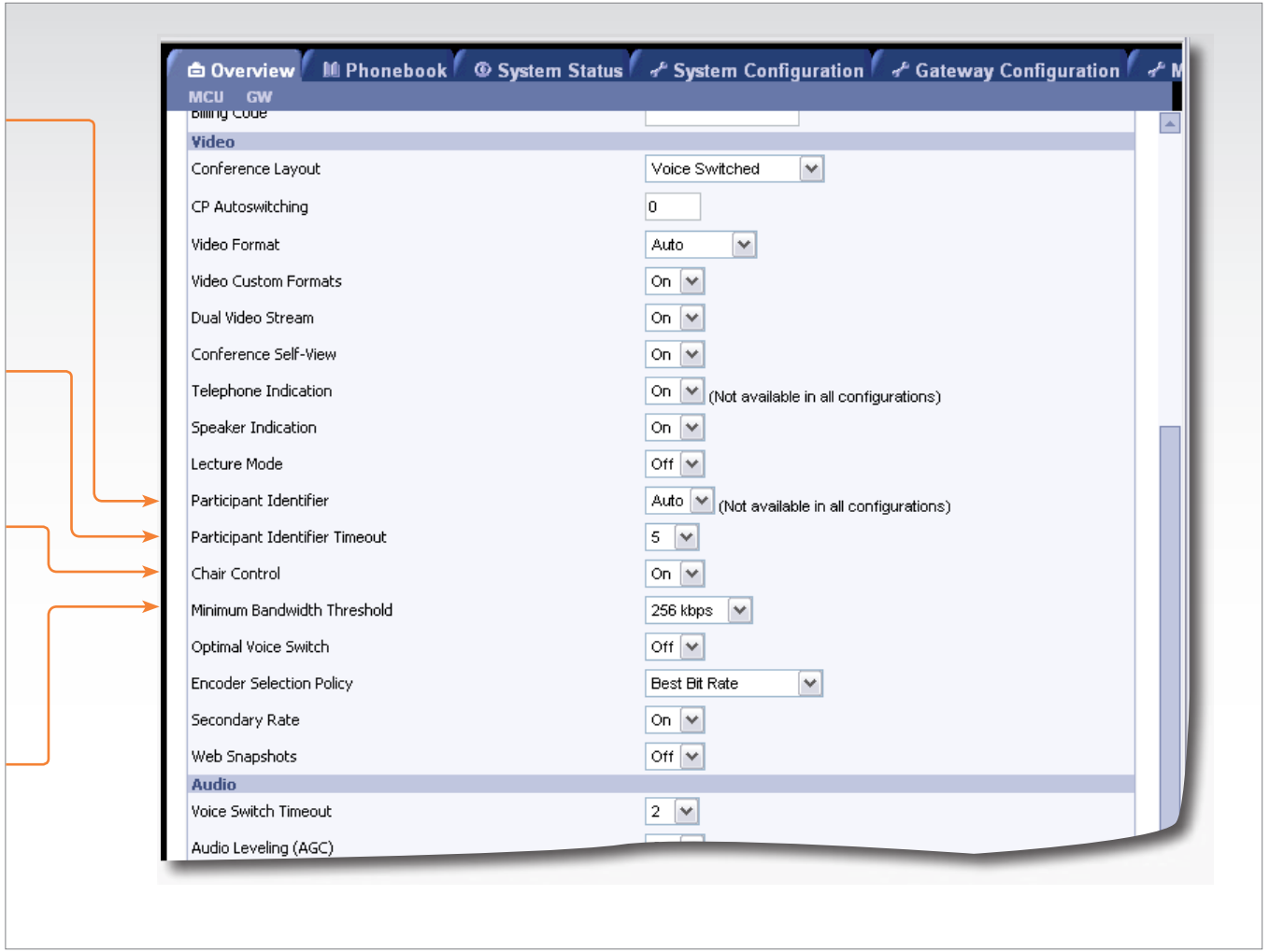
Off: Set to Off to disable Chair Control,

Minimum Bandwidth Threshold

If a participant calls in with a lower bandwidth than the Minimum Bandwidth Threshold, the participant will receive audio only (not live video) as well as a poster saying the bandwidth is to too low. After 10 seconds the participant will receive low rate video.

- The Minimum Bandwidth Threshold can be modified during a conference.
- The system will move calls below the defined Minimum Bandwidth Threshold to a low rate encoder.

NOTE: Once a participant is moved to the low rate encoder, they will not be moved back even if the Minimum Bandwidth Threshold is lowered.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Video settings

Optimal Voice Switch

On: Set to On to enable Optimal video format in Voice Switch mode, if the connected endpoints allows this.

Off: When set to Off there will be normal transcoding when doing Voice switch.

1. Optimal Voice Switch is only available on IP.
2. Icons and text are not available when set to On.

Encoder Selection Policy

Best Bit Rate: Set to Best Bit Rate to make the MPS prioritize the video quality for sites based on bit rate.
The system will move participants with a Low Video Rate to a secondary encoder, if it is available. If no sites are moved, the system will move sites with Low Video Standard.

Best Video Standard: Set to Best Video Standard to make the MPS prioritize sites based on video standard.
The system will move participants with a Low Video Standard to a secondary encoder, if it is available. If no sites are moved, the system will move sites with Low Video Rate.

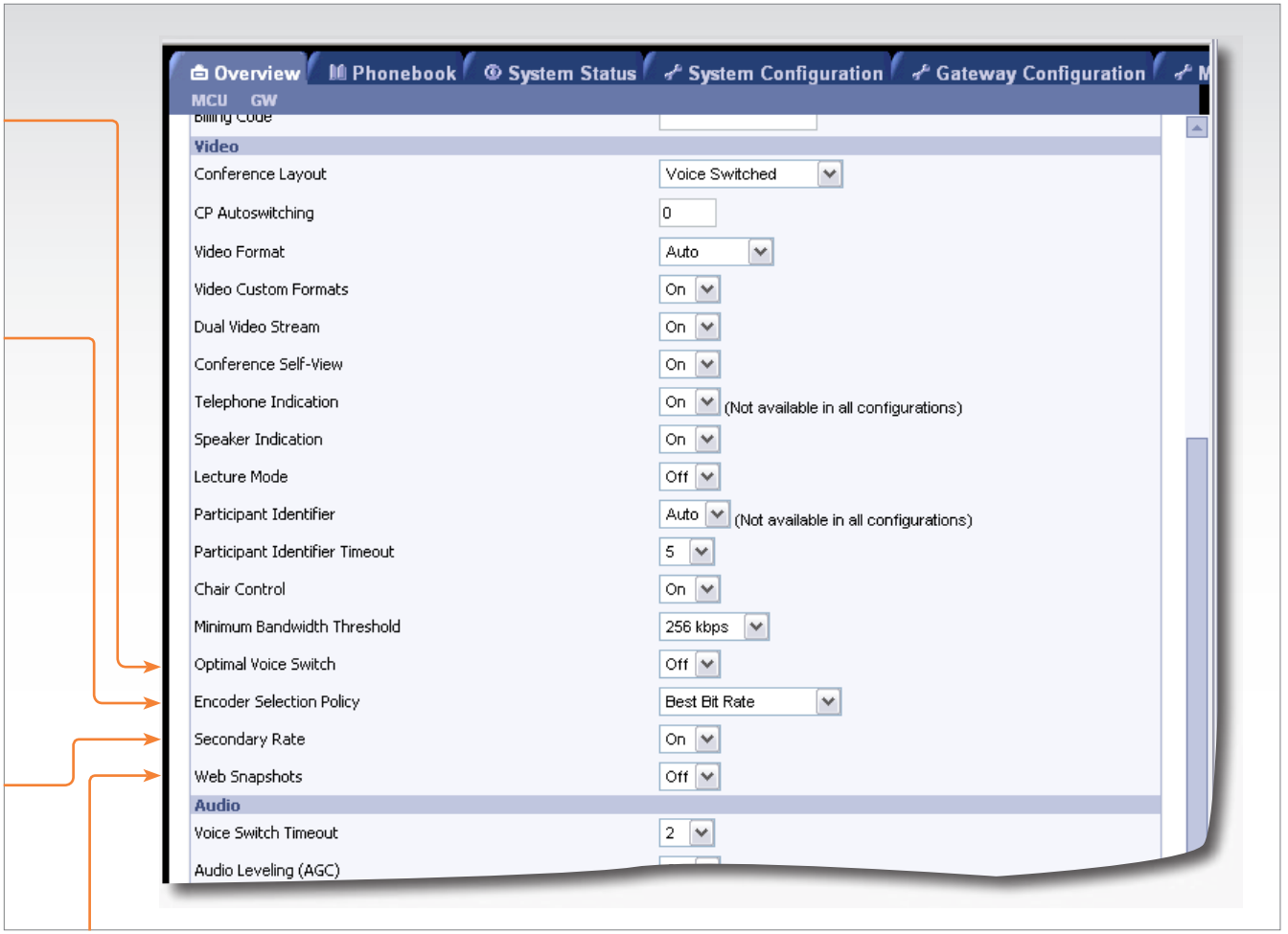
Best Resolution: Set to Best Resolution to make the MPS prioritize the video quality for sites based on resolution.

The system will move participants with a Low Resolution to a secondary encoder, if it is available. If no sites are moved, the system will move sites with low video rate.

Secondary Rate

On: Set to On to enable Secondary Rate. The conference will support two outgoing bandwidths if needed, in addition to the low rate video.

Off: Set to Off to disable Secondary Rate.



Web Snapshots

The web snapshots are shown in the upper right corner of the web interface, and will show snapshots of the video from the participants and dual video stream. The snapshots are updated in accordance to the refresh rate (placed above the snapshot).

On: Set to On to enable Web Snapshots. When set to On the Conference Snapshot and Dual Video Stream Snapshot will show the video transmitted from the MCU to the participants.

Off: When set to Off a picture will appear telling that the Web Snapshots are disabled.

Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Audio settings

Voice Switch Timeout

Defines the number of seconds between 1 and 10, a participant must speak before it gets the speaker indication and is shown as the speaker to the other endpoints.

A long timeout might be suitable in noisy environments and in conferences with many participants.

Audio Leveling (AGC)

Ensures that all participants will receive the same audio level from all other participants, regardless of the levels transmitted. AGC - Automatic Gain Control.

In most conferences, the participants will speak at different levels. As a result, some of the participants are harder to hear than others. The Audio Leveling corrects this problem by automatically increasing the microphone levels when “quiet” or “distant” people speak, and by decreasing the microphone levels when “louder” people speak.

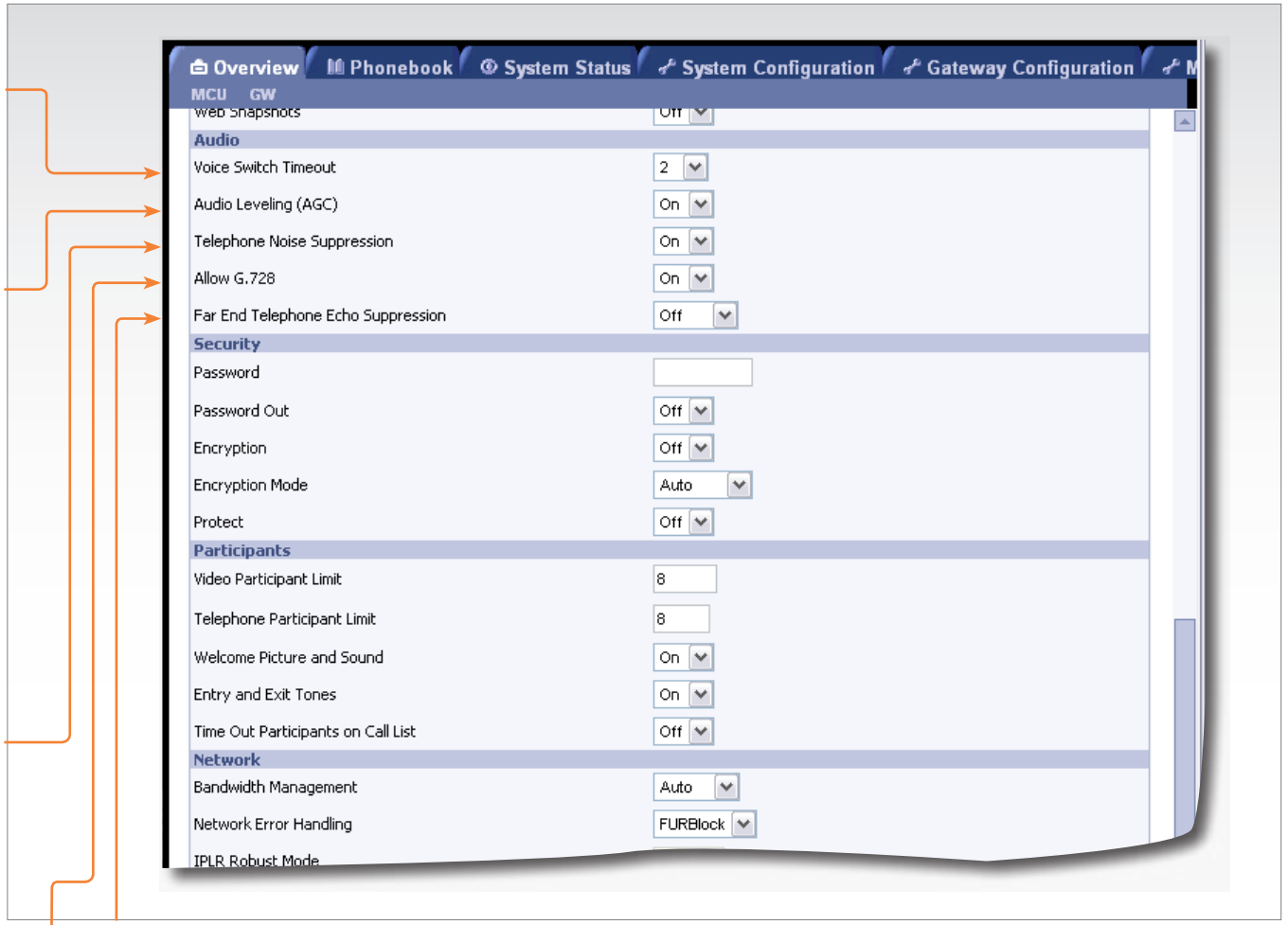
On: When set to On the MCU maintains the audio signal level at a fixed value by attenuating strong signals and amplifying weak signals. Very weak signals, i.e. noise alone, will not be amplified.

Off: Set to Off to disable Audio Leveling (AGC).

Telephone Noise Suppression

On: Set to On to enable Telephone Noise Suppression. Attenuates the noise which normally is introduced when adding mobile phones to a conference. The background noise normally heard when the telephone participant is not speaking will be attenuated.

Off: Set to Off to disable Telephone Noise Suppression.



Allow G.728

On: The MCU supports high quality audio even on low call rate. On low call rate the MCU will prioritize G.722.1. The video participants which do not support G.722.1 will receive low quality audio G.728 instead when Allow G.728 is set to On.

Off: To ensure high quality audio on low call rate, set Allow G.728 to Off. Then video participants which are not able to support G.722.1, will receive G.722 instead.

Far End Telephone Echo Suppression

Analog telephone lines, speaker phones and telephone headsets may all cause echo. The Far End Telephone Echo Suppression function eliminates some or all of the experienced echo.

Off: Set to Off to disable Far End Telephone Echo Suppression.

Normal: Set to Normal to remove weak echo.

High: Set to High to remove strong echo.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Security settings

Password

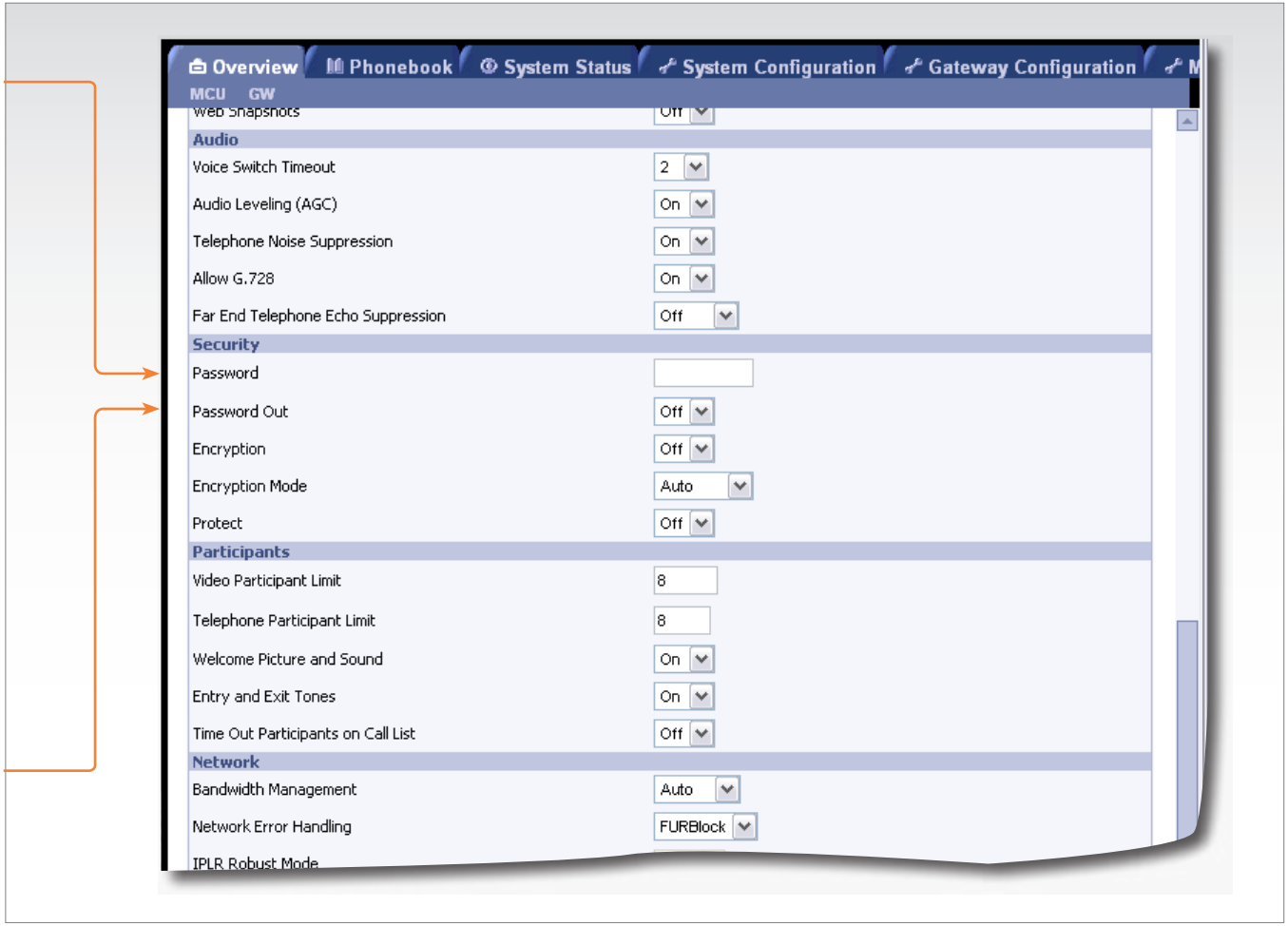
To ensure only authorized participants are able to join this conference you can set a password. Then the participants must enter the password to join this conference. The password can be numerical only.

- When dialling into a password protected conference, the participant is met with the 'Password Enquiry' screen and sound, asking the participant to enter a password. This can be performed via a menu generated by the videoconferencing system (H.243 Password) or via DTMF (telephone) tones.
- Until the correct password is entered, the participant will not be able to hear or see any of the other participants. After entering the correct password and confirming (typically by pressing 'OK' or the hash key), the participant will join the conference.
- Should the password be incorrect, the participant is met with the 'Password Incorrect' screen and after a few seconds, the 'Password Enquiry' screen and sound appear again. If the participant enters a wrong password three times, the participant will be disconnected.
- With no password entered in this field, participants can join the conference without entering a password

Password Out

On: When set to On and dialling out from a password protected conference, the participant is met with the 'Password Enquiry' screen and sound, asking the participant to enter a password. This setting can be used to ensure that only authorized participants are able to join the conference also when dialling out from the conference.

Off: When set to Off no password is required when dialling out.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Security settings

Encryption

On: When set to On all participants in the conference must support DES or AES encryption (available on all TANDBERG endpoints using software version B4.0 or later). Participants not supporting encryption will be shown the 'Encryption Required' screen for 60 seconds before they are disconnected from the conference.

Off: When set to Off the conference is not encrypted.

In an encrypted conference, there is no support for telephone participants.

Encryption Mode

This settings only applies if Encryption is set to On (see above).

Auto: Set to Auto to use the highest level of encryption available on each of the participants connected in the conference. This means that there can be a mix of DES and AES encrypted connections in the same conference.

AES-128: Set to AES-128 to allow only participants with AES 128 bit encryption capabilities. Participants without this capability will not be able to join the conference.

DES: Set to DES to allow only participants with DES 56 bit encryption capabilities. Participants without this capability will not be able to join the conference.

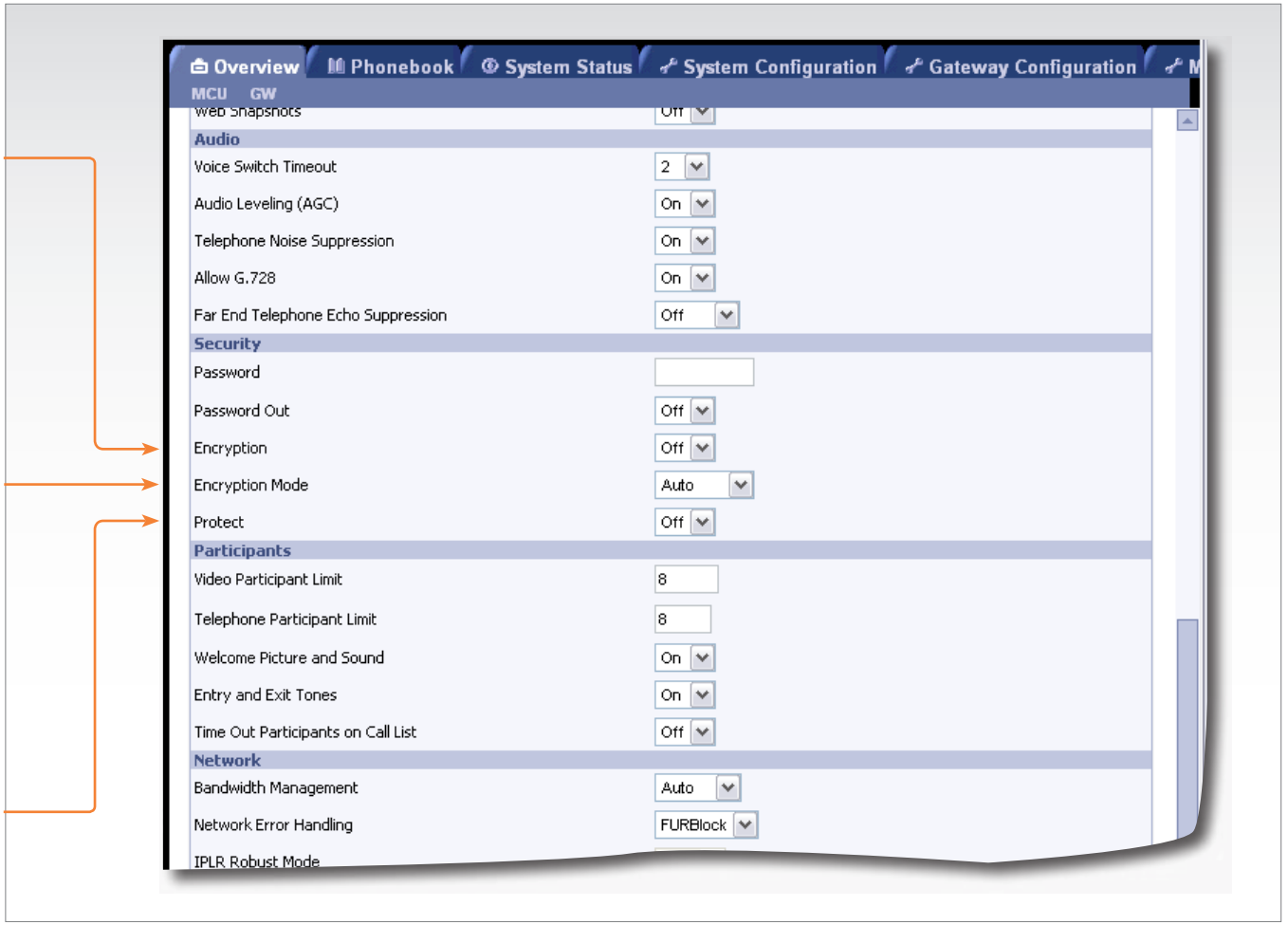
Protect

On: When Protect mode is set to On:

1. Only predefined Protected Numbers are allowed to join this conference.
2. The Protected Numbers field will be shown, and Protected Numbers can be configured from the Dial In Configuration in the MCU Conference Overview page.

Off: Set to Off to disable the Protect mode.

For further information on Protected Numbers, see Protected Numbers in the Dial In Configuration in the Manage an Ongoing Conference section in this book.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Participants settings

Video Participant Limit

Defines the maximum number of Video Participants allowed in the conference and reserves the number of needed Advanced Video Option ports for this conference.

Values: 0 - 40 for MPS 200 and 0 - 160 for MPS 800.

Telephone Participant Limit

Defines the maximum number of Telephone Participants allowed in the conference.

Values: 0 - 32 for MPS 200 and 0 - 48 for MPS 800.

Welcome Picture and Sound

On: When set to On a Welcome screen and audio message will be shown to each new participant of the conference.

Off: Set to Off to disable the Welcome screen and audio message.

Entry and Exit Tones

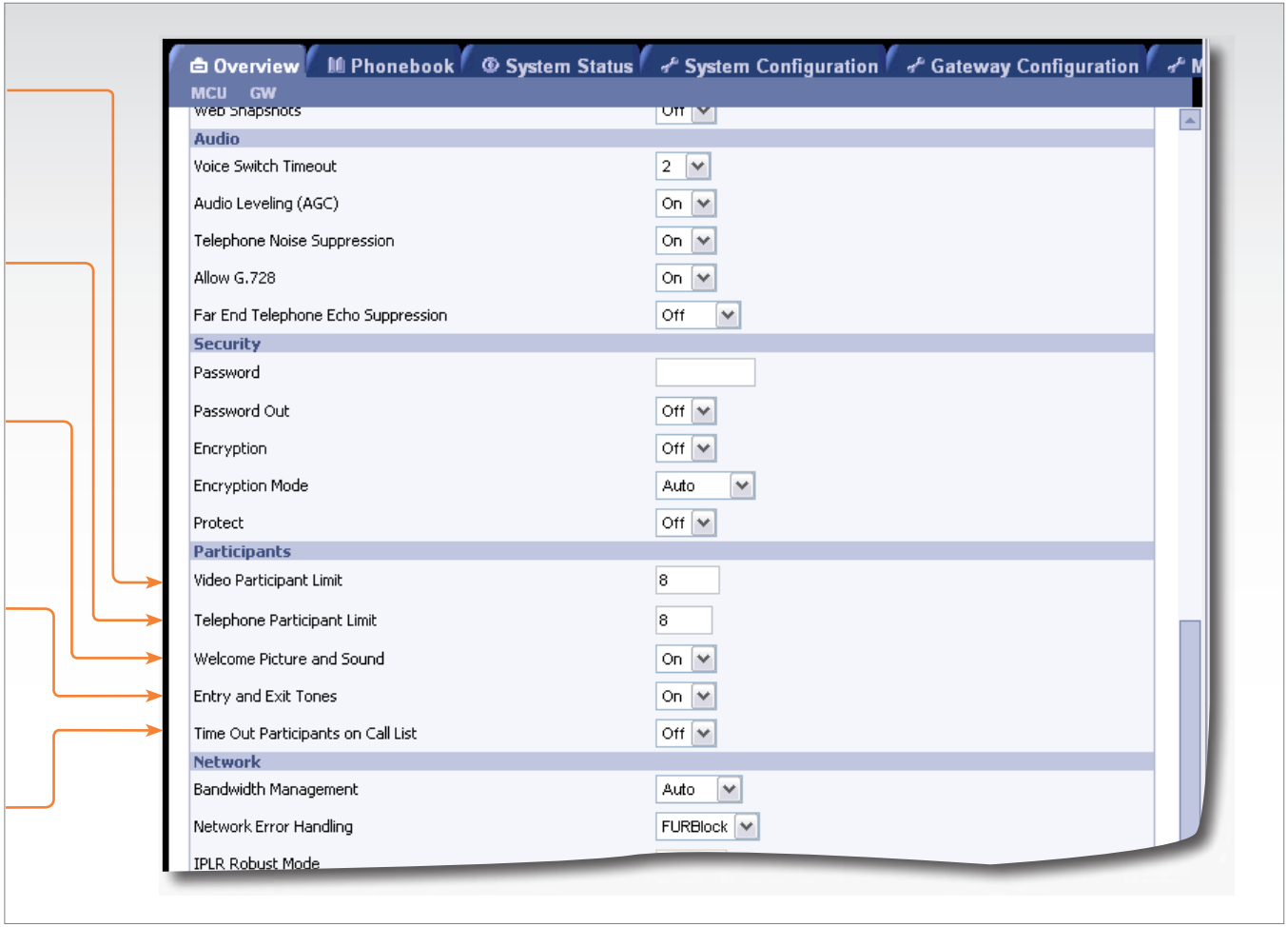
On: When set to On a tone signal will be heard each time a participant is entering or leaving the conference.

Off: Set to Off to disable the Entry and Exit Tones.

Timeout Participants from Call List

On: When set to On all participants that have been disconnected from the conference will be cleared from the Call List within 2 minutes.

Off: Set to Off to disable the Timeout Participants from Call List.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

Select Cancel to discard all changes and return to the Conference Overview page.

Create conference, *cont...*

Network settings

Bandwidth Management

Manual: Disables automatic regulations of sites to Low rate encoder, based on video rate reports.

Auto: Enables automatic regulations of sites to Low rate encoder, based on video rate reports.

Network Error Handling

None: Set to None to not enable error handling.

IPLR: Set to IPLR (Intelligent Packet Loss Recovery) if one or more sites are experiencing network errors.

FURBlock: Set to FURBlock (Fast Update Request Block) if one or more sites are experiencing network errors.

The Network Error Handling may be set to IPLR (Intelligent Packet Loss Recovery) or FURBlock (Fast Update Request Block) if one or more sites are experiencing network errors.

IPLR Robust Mode

Auto: When set to Auto, the IPLR Robust Mode is turned on for each encoder when needed.

On: When set to On, the IPLR Robust Mode is on for all encoders.

Please refer to Intelligent Packet Loss Recovery (IPLR) in the Appendices section.

FUR Block Sites

Auto: When set to Auto, the FUR's from sites that send too many will be blocked.

On: When set to On, the FUR's from all sites will be blocked.

FUR Filter Interval

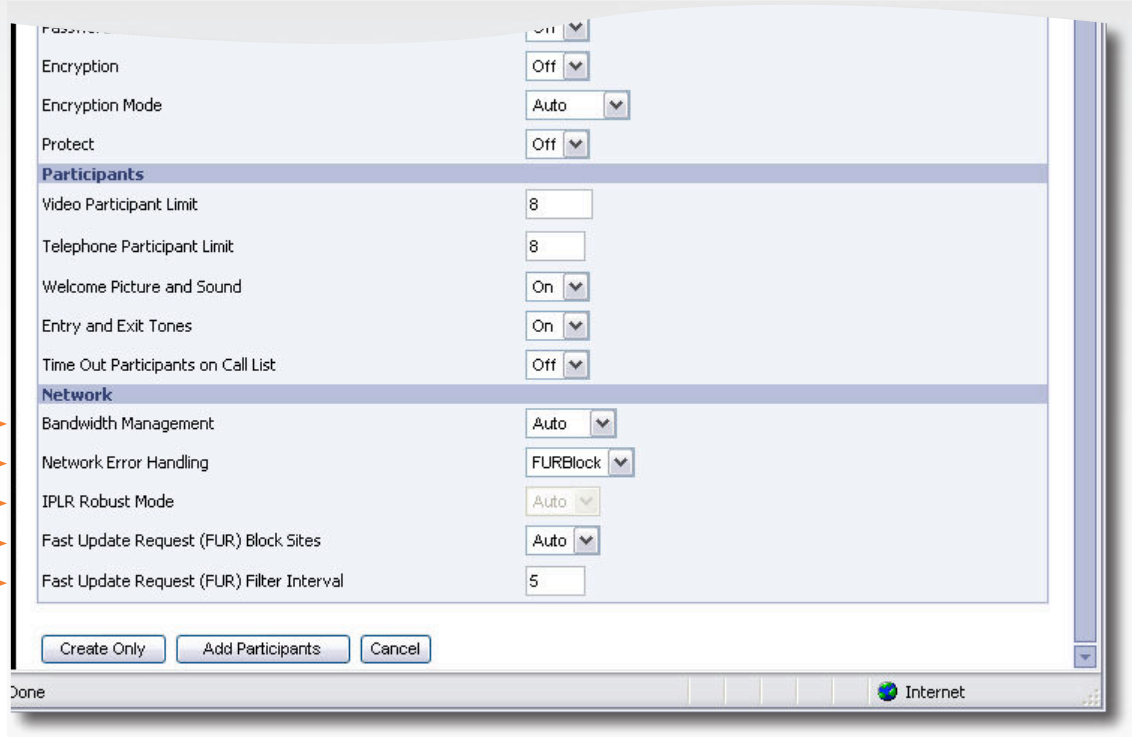
Denotes the number of seconds between FUR's, e.g. the minimum time between FURs that will refresh the picture.

Network

If a conference participant is experiencing poor network quality it will send Fast Update Requests (FUR) to the encoder in the MCU to make it refresh the picture. This can be observed as a short flash in the picture.

Poor network conditions for one participant may have a deteriorating effect on the video quality for some of the participants in the conference.

In an effort to reduce this effect the Network Error Handling can be used.



Create Only

Select Create Only to create a conference, with the above-specified configuration, without dialling out to any participants.

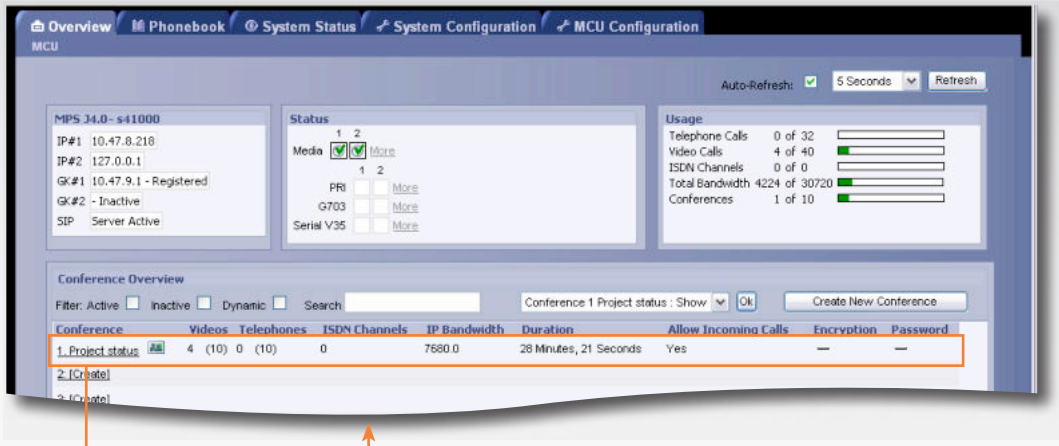
Add Participants

Select Add Participants to add participants or to manually dial a participant.

Cancel

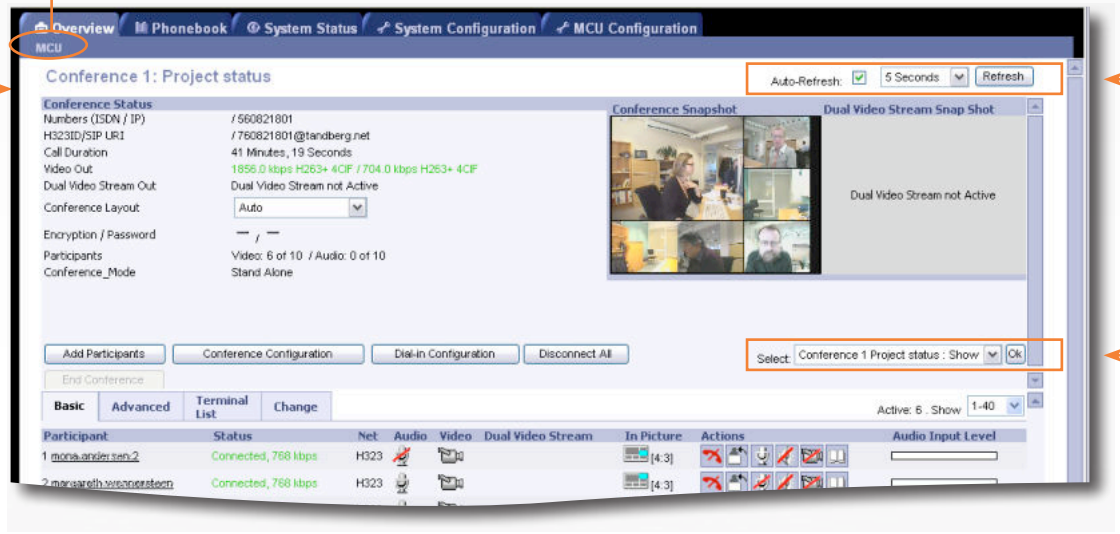
Select Cancel to discard all changes and return to the Conference Overview page.

Manage an active conference



Frequently used numbers can be stored in the Phone Book or a management system, e.g. TANDBERG Management Suite (TMS).

2 Click on MCU to return to the Conference Overview.



1 Click on a conference in the list to see the Conference Status for this conference.

The Conference Status views are described in detail on the following pages.

Auto Refresh: Check/uncheck the Auto Refresh to put it On/Off.

Auto Refresh Rate: Select the desired Refresh Rate for Auto Refresh.

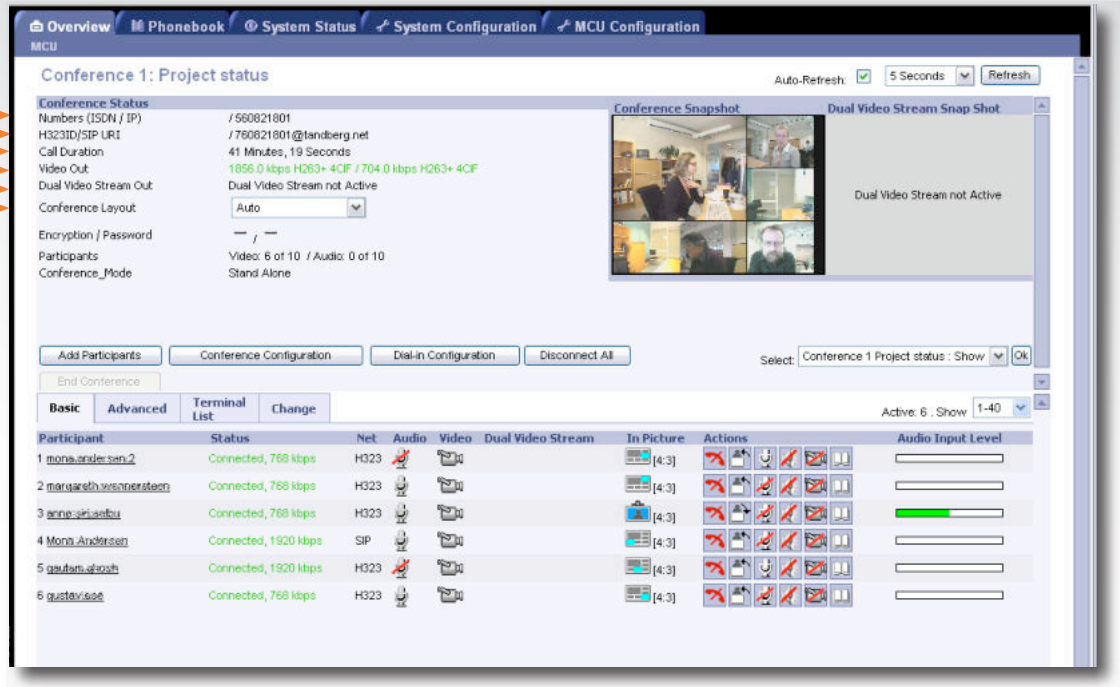
Refresh: Push the Refresh button to manually refresh the page.

Select: Select a conference from the list and press the OK button to change the view to show the selected conference.

Overview MCU

Conference Status

The Conference Status page shows information about a conference and lets you control the conference.



Numbers

Shows the dial in number for the conference, on ISDN or IP.

Each conference has separate dial in numbers.

H.323 / SIP URI

Shows the dial in address for the conference, on H.323 or SIP.

Call Duration

Shows the call duration of the current conference.

If specified, the Maximum Call Duration for the conference, is also shown. The Maximum Call Duration is set when you Create Conference or in the Conference Template Configuration

Video Out

Shows the outgoing video rate, the video encoding algorithm and the resolution that is transmitted from the MCU to the participants.

Dual Video Stream Out

If one of the participants is transmitting DuoVideo^{TF}/H.239/BFCP, the outgoing Dual Video Stream rate, the Dual Video Stream encoding algorithm and the resolution is shown here.

Participants not capable of receiving Dual Video Stream, will only receive the main video.

Conference Layout

Shows the selected conference layout. The following layouts are available:

Auto, Voice Switched, 2 Split, 3 Split, 4 Split, 4+3 Split, 5+1 Split, 7+1 Split, 8+2 Split Top-Bottom, 8+2 Split Bottom-Top, 8+2 Split Top, 8+2 Split Bottom, 9 Split, 12+1 Split Center, 12+1 Split Top left, 12+2 Split, 16 Split, 30+2 Split, 2+1 Split Wide, 3 Split Wide, 3+1 Split Wide, 4+1 Split Wide, 6 Split Wide, 8+1 Split Wide, 12 Split Wide and CP Auto.

The Conference Layout can also be set when you Create Conference and in the Conference Template Configuration.

Overview MCU, *cont...*

Conference Status

The Conference Status page shows information about a conference and lets you control the conference.

Encryption / Password

Encryption: Shows what type of encryption is allowed for the conference.

Password: Shows the key symbol if a dial in password has been assigned to the conference. If no key symbol is present, the conference is not password protected.

Participants

Video / Telephone: Shows the current number of video and telephone participants in the conference and the maximum allowed.

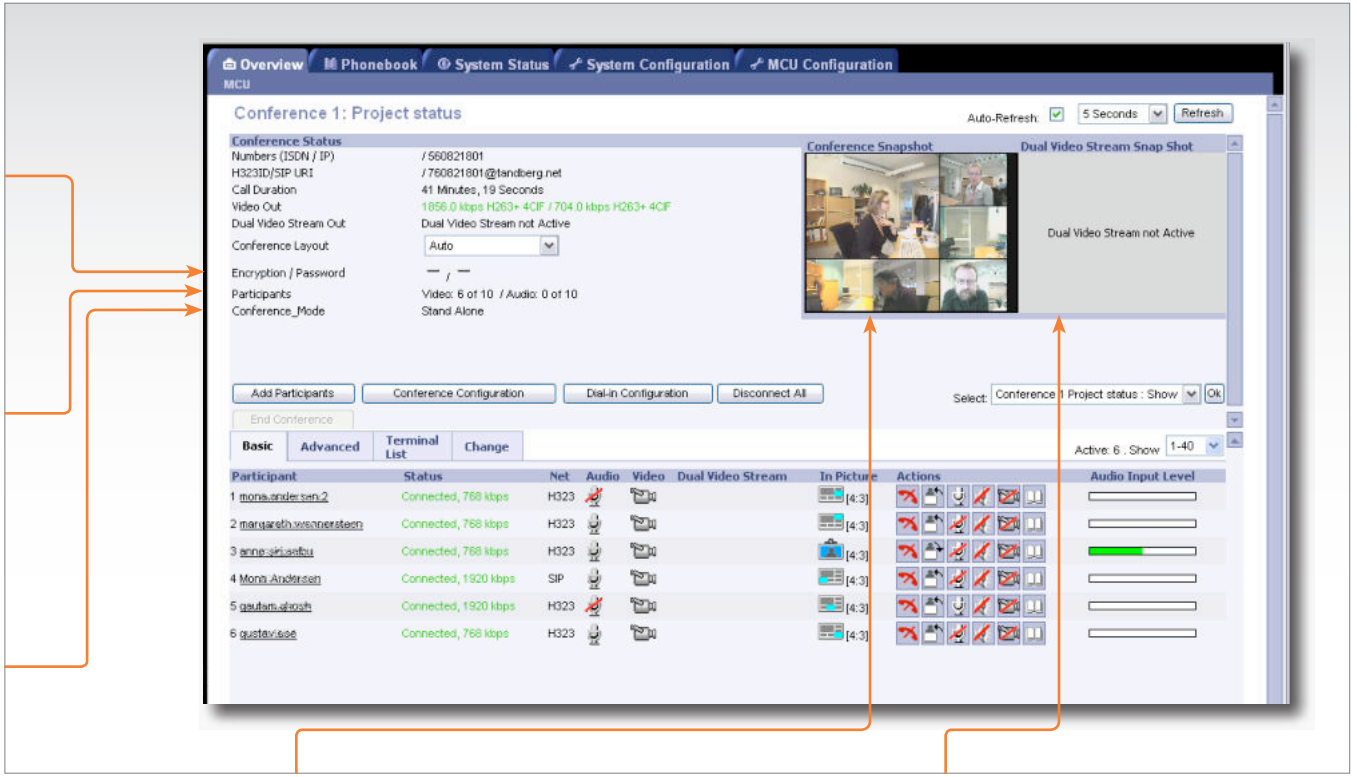
In the example to the right there are 6 of 10 participants on video and 0 of 10 participants on audio.

Conference Mode

Stand Alone: This mode indicates that a normal conference is active.

Cascading Master: Indicates that this conference has become Master when connecting to another MCU. If the `<Warning: Multiple masters, irregular behaviour expected>` is seen, more than one conference has been forced to Master, which is not recommended.

Cascading Slave: Indicates that this conference has become Slave when connecting to another MCU



Conference Snapshot

Conference Snapshot Shows a snapshot of the video transmitted from the MCU to the participants.

Click on the picture to enlarge it in a separate window.

In an encrypted conference, Conference Snapshots are not available.

Dual Video Stream Snapshot

Snapshot Shows a snapshot of the Dual Video Stream transmitted from the MCU to the participants.

Click on the picture to enlarge it in a separate window.

Overview MCU, *cont...*

Add Participants

The Conference Status page shows information about a conference and lets you control the conference.

Add Participants

To add new participants to the conference, press the Add Participants button. The Call Participants window will be shown.

Using the Phone Book

Select an entry from the phone book and press the Call Participants button to call the participant.

Search in Phone Book

Enter a name in the Find field to search for a name in the phone book.

Modify a Phone Book Entry

Use Copy entry to Manual Dial to modify an existing entry in the phone book.

Add Several Participants

To add several participants at the same time, select an entry from the phone book or fill in the manual dial area and press the Add --> button to add the participant to the Participant list. Press the Call Participants button to call the participants.

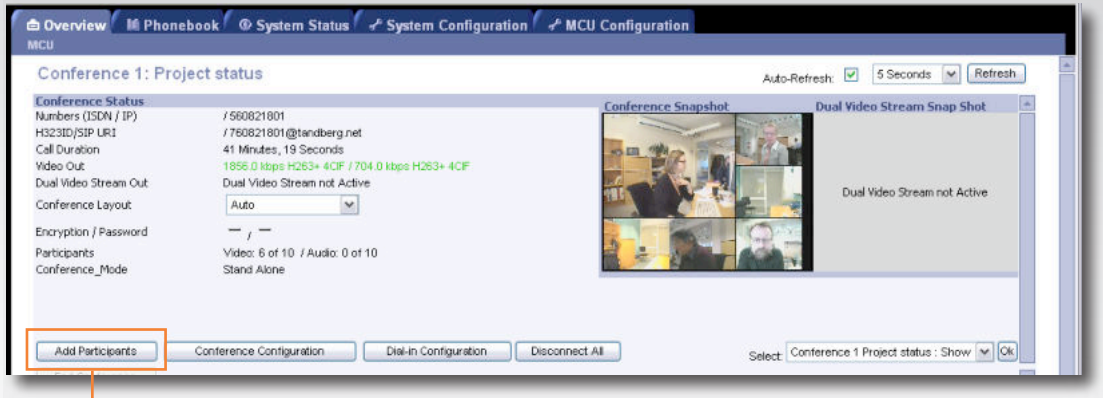
Remove Participants from the Participants List

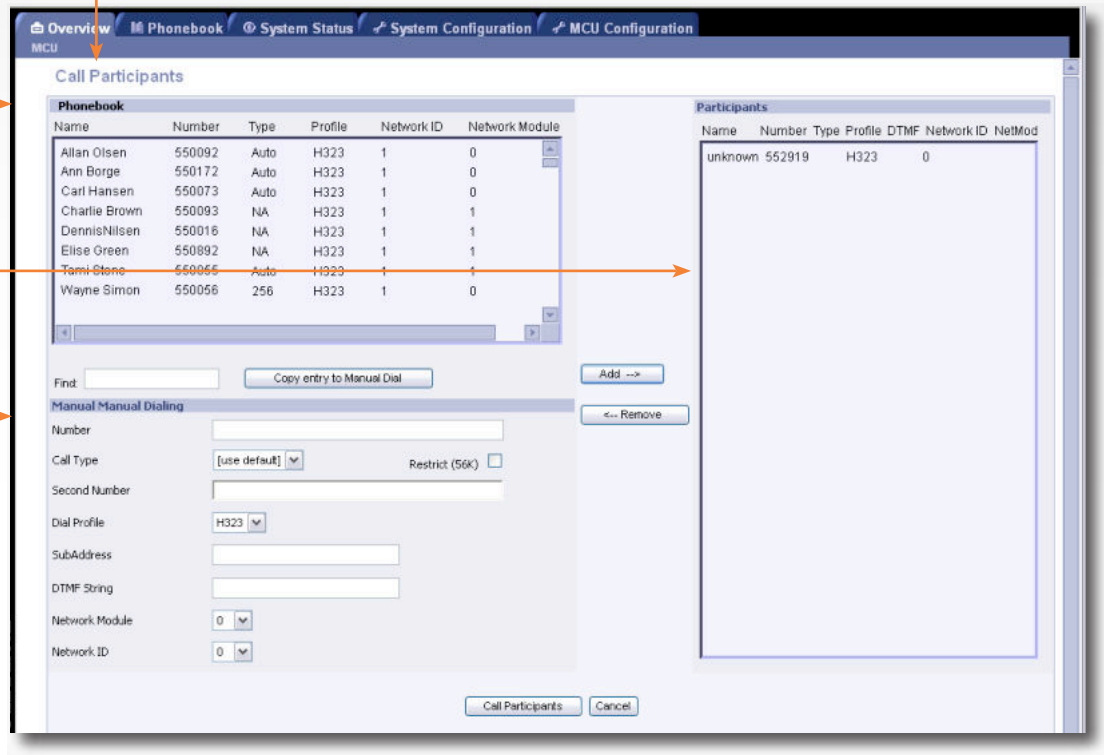
To remove a participant from the Participants List, select an entry from the list and press the <-- Remove button.

Using the Manual Dialling

If a participant is not listed in the phone book, use the Manual Dial area and press the Call Participants button to call the participant.

The Manual Dialling is explained on the next page.





Overview MCU, *cont...*

Add Participants

The Conference Status page shows information about a conference and lets you control the conference.

Add Participants

To add new participants to the conference, press the Add Participants button. The Call Participants window will be shown.

Number

Enter the video number, telephone number or RS366 dial string.

For G.703 Leased Line calls, this is the call number (1 to 5).

Call Type

Select the call rate to be used.

Use Default (default) is the configured Maximum Conferense Rate.

Custom Selection: Use Default, 1 x H.221, 2 x H.221, 64 kbps, 128 kbps, 192 kbps, 256 kbps, 320 kbps, 384kbps, H0, 512 kbps, 768 kbps, 1152 kbps, 1472 kbps, 1536 kbps, 1920 kbps and Telephone

Example: When calling an endpoint on a V.35 port configured for 384 kbps and the default call rate is 768 kbps, you must select 384 kbps to ensure proper framing and syncing.

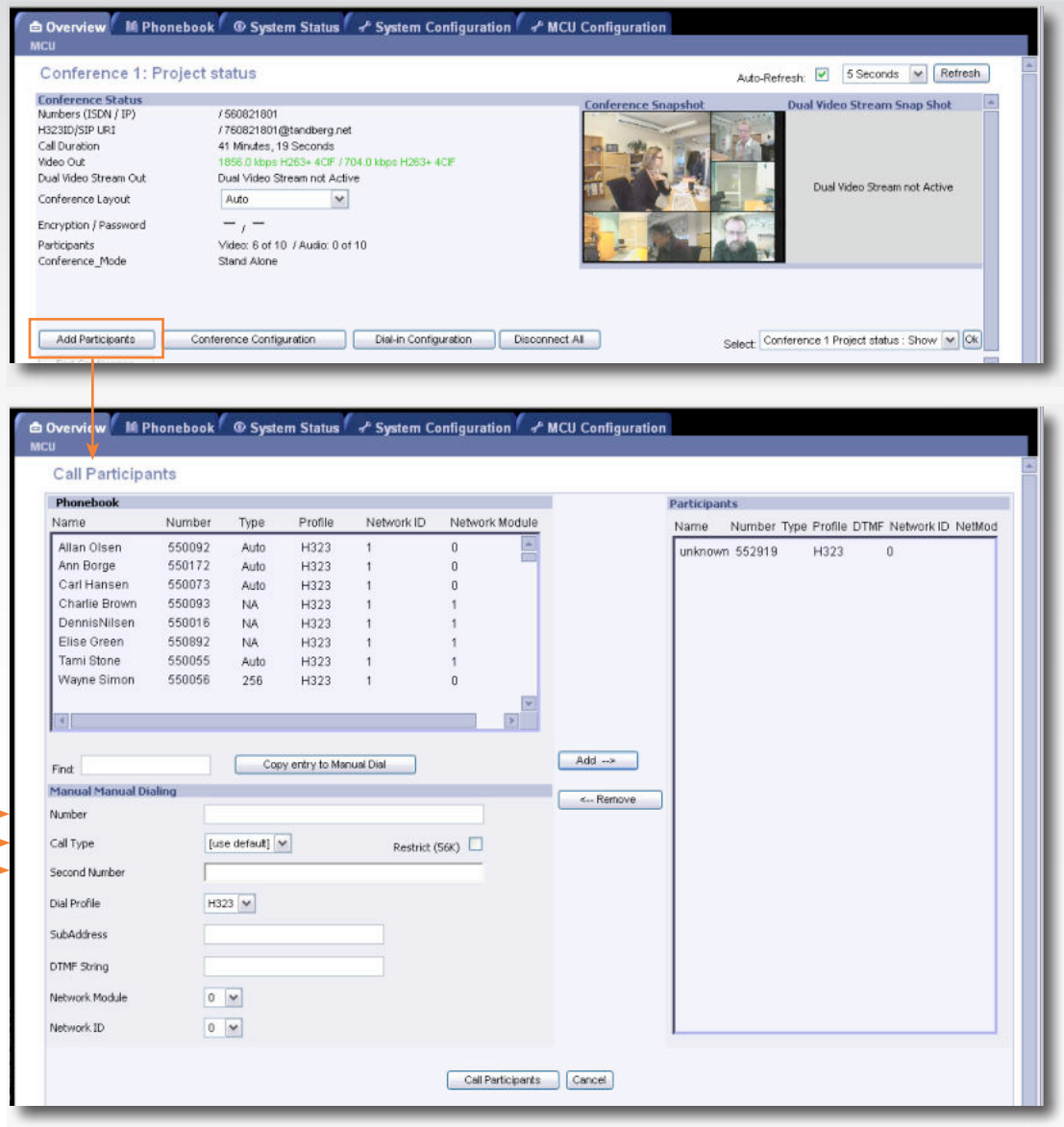
Restrict (56K)

Select Restrict (56K) to use 56 kbps per ISDN B-channel.

Second Number

If two numbers are required, both should be specified for 2x64 kbps and 2x56 kbps calls.

Leave blank (default) if the endpoint only has one number.



Overview MCU, *cont...*

Add Participants

The Conference Status page shows information about a conference and lets you control the conference.

Dial Profile

Select which dial profile to use for this phone book entry.

Auto: The MPS will automatically select the appropriate profile

ISDN: ISDN will always be used for dialling.

H.323: IP will always be used when dialling.

SIP: SIP will always be used when dialling.

If defined, custom Dial Profiles (Network Profiles) will show in the list.

Subaddress

The Subaddress is used to address different participants on the same ISDN line or TCS4 dialling.

DTMF String

Specify the DTMF (Dual Tone Multi-Frequency) or Touch Tone string.

Network Module

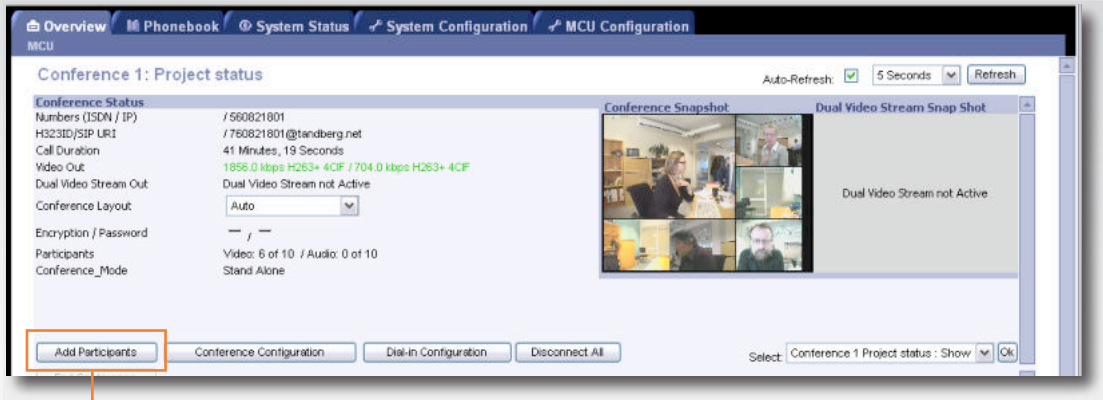
Specify which Network Interface Card to use for outgoing calls. Enter a value between 0 and 6.

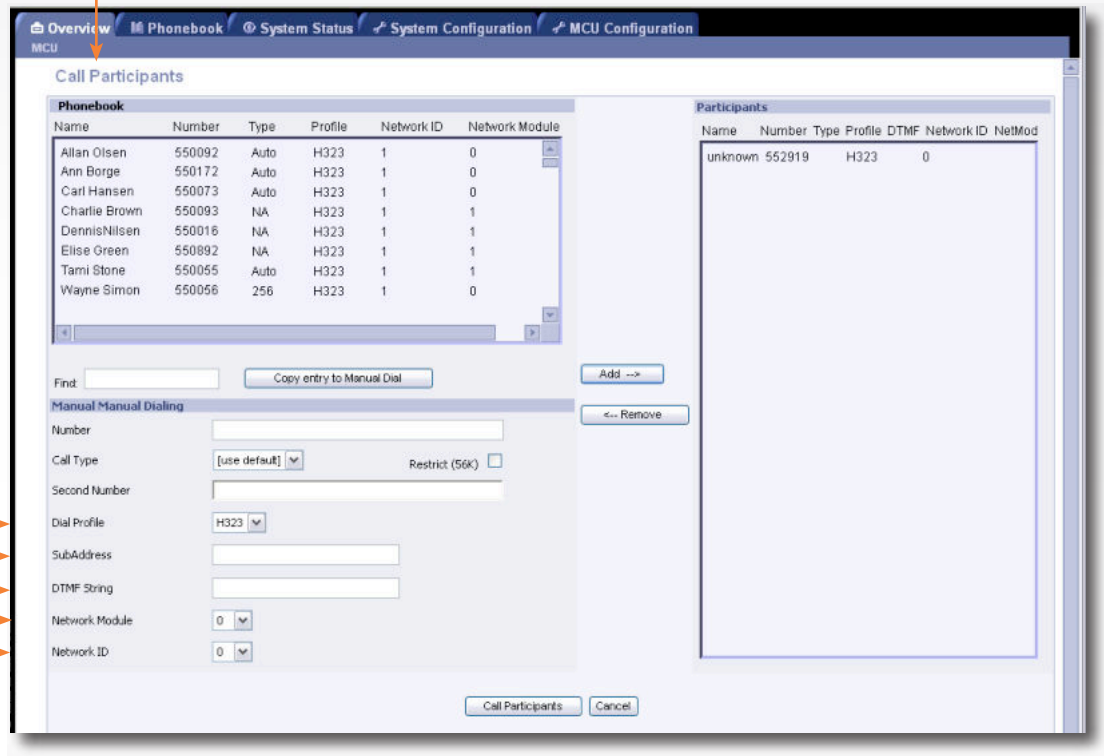
- Optional for ISDN calls
- Mandatory for V.35 calls and G.703 Leased Line calls.

Network ID

Used to identify port or interface number within a network module. Enter a value between 1 and 32:

- Specify which IP network to use, only 1 and 2 are valid values (optional).
- Specify which V.35 port to use (mandatory).





Add Participants - Examples

Example with H.323/IP (H.323 ID):

The H.323 ID must be registered to a Gate-keeper.

Number: sales.manager
Call Type: N/A
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: H.323 or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: N/A
Network ID: N/A

Example with H.323/IP (IP Address):

Calling a video system directly by IP Address.

Number: 127.0.0.16
Call Type: N/A
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: H.323 or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: N/A
Network ID: N/A

Example with ISDN:

A prefix might be required by the ISDN switch.

Number: 067125125
Call Type: N/A
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: ISDN or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: N/A
Network ID: N/A

Example with G.703 Leased Line:

Find which Slot and Port to use for the Network Interface (ISDN) to the call.

Number: 3 (Call number 3, of max 5)
Call Type: 256 kbps (the # of channels x 64 kbps)
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: ISDN or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: 2 (Network Interface Card (ISDN) in slot 2)
Network ID: 5 (Port 5, PRI number 5)

Example with V.35:

Find which Slot and Port to use for the Network Interface (V.35) to the call.

Number: Leave empty
Call Type: 768 kbps (the # of channels x 64 kbps)
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: ISDN or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: 3 (Network Interface Card (V.35) in slot 3)
Network ID: 14 (Port 14)

Example with V.35 RS366:

Find which Network Interface (V.35) to use for the call.

Number: 6700#4#6#0
Call Type: 384 kbps (the # of channels x 64 kbps)
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: ISDN or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: 4 (Network Interface Card (V.35) in slot 4)
Network ID: 31 (Port 31)

Example with SIP:

The SIP Alias must be registered to a SIP Server.

Number: SIP:j.doe@sip.tandberg.net
Call Type: N/A
Restrict (56 kbps): N/A
Second Number: N/A
Dial Profile: SIP or Auto
Subaddress: N/A
DTMF String: N/A
Network Module: N/A
Network ID: N/A

Overview MCU, *cont...*

Edit Conference Configuration

The Conference Status page shows information about a conference and lets you control the conference.

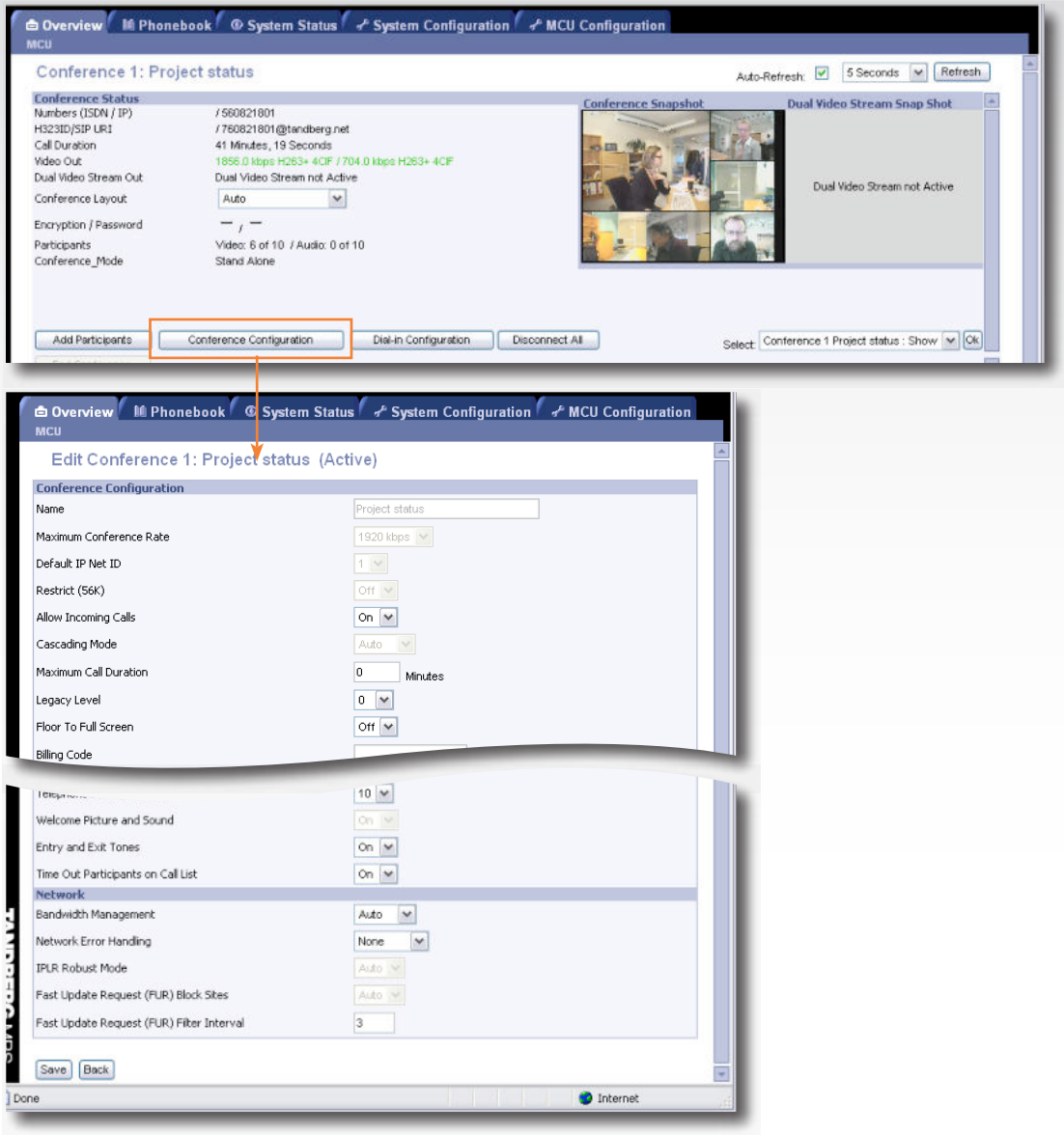
Conference Configuration

To change the conference configuration for this conference, press the Conference Configuration button to show the Edit Conference window.

Some settings cannot be changed when a conference is active.

Read More

To see a description of each setting go to the MCU Overview > Create Conference section.



Overview MCU, *cont...*

Dial-In Configuration

The Conference Status page shows information about a conference and lets you control the conference.

Dial In Configuration

To specify a dynamic dial in number for a conference, press the Dial In Configuration button. The Direct Dial In Number to Conference # window will be shown.

Using Dial In Direct Numbers

To use Dial In Direct (DID) numbers, a DID has to be configured for the MPS. For details see Dial In Direct Numbers in the MCU Configuration > Static Conferences section.

Numbers

The DID (Dial In Direct) numbers must be pre-defined MPS dial in numbers, either available on the ISDN PRI or available by H.323 prefixes.

Number: Specify the dial in number that will be directed into this conference.

Network Profile: Select which net profile to use.

Caller Id: Specify the caller id, given by the network, for the calling participant. If the caller id does not match, the participant will not be included in the conference. If the Caller Id is left blank, no check will be performed.

Protected Numbers

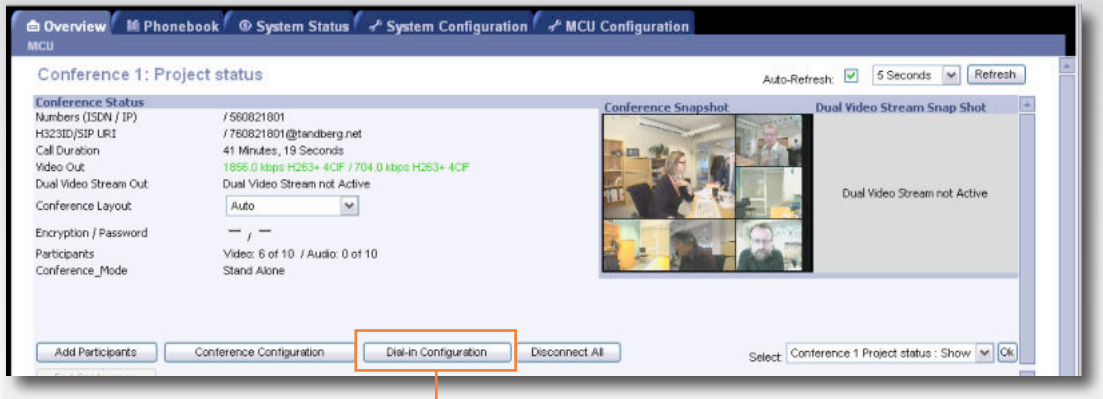
If the Protect setting is set to On:

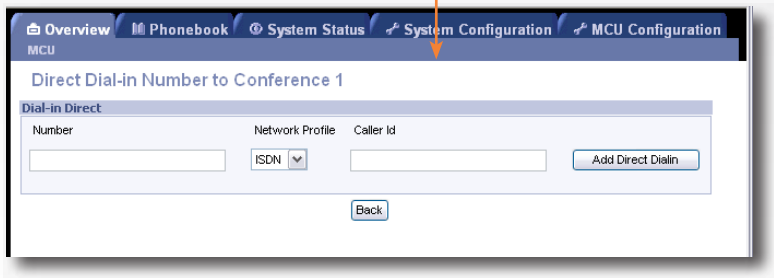
1. The Protected Number field will be shown
2. Only Protected Numbers are allowed to join this conference

Number: Specify the number given by the network for the calling participant.

Network Profile: Specify which net profile to use.

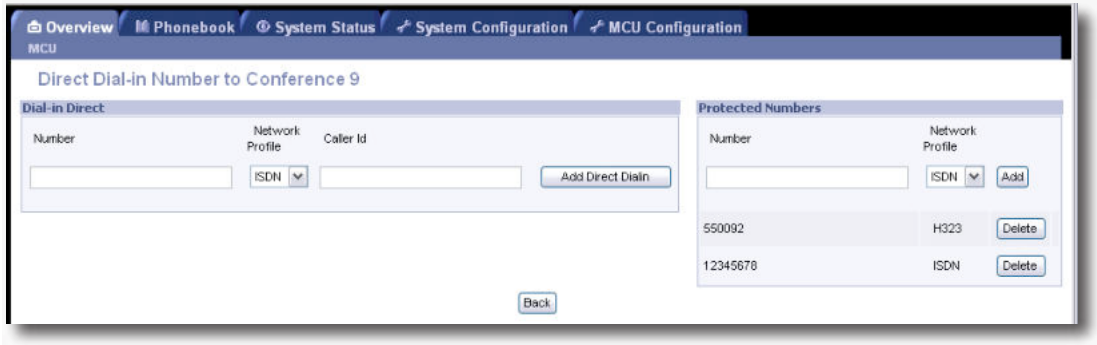
For further information, see the Protect setting in the Overview > MCU > Create Conference section.





The Protect setting is set to Off (default) and the Protected Number field is not shown.

The Protect setting is set to On and the Protected Number field is shown. Only the Protected Numbers configured in this dialog will be allowed to join this conference.



Overview MCU, *cont...*

Conference status, the Basic tab

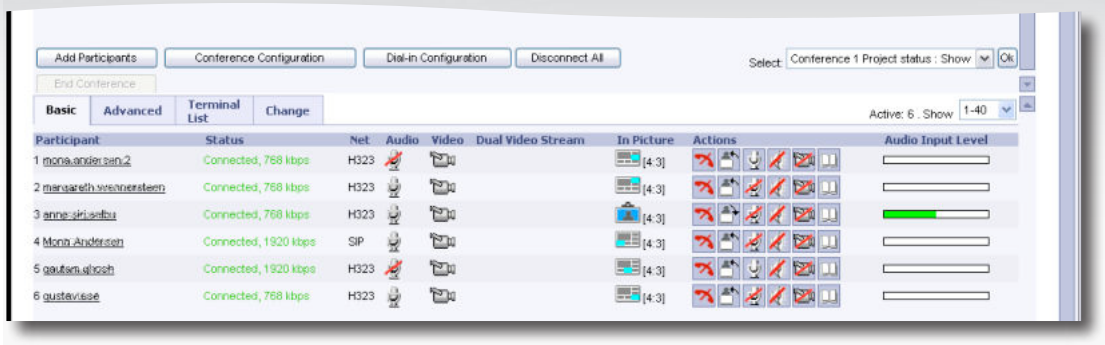
The Conference Status page shows information about a conference and lets you control the conference.

Basic View - Provides a basic overview of all the participants in the conference.

Advanced View - Gives detailed information on the audio in, audio out and video in protocols and the bandwidth used by these protocols.

Terminal List - List the participants in the conference. If the conference is cascaded to another conference, this will be shown here.

Change - The change view is used to modify endpoint constraints or move the endpoint to another conference.



Disconnect One Participant

To disconnect a participants in the conference

1. Select a participant from the list and press the red Disconnect button.
2. A confirmation window will be shown. Press OK to disconnect the participant.

Disconnect All

To disconnect all participants in the conference:

1. Press the Disconnect All button.
2. A confirmation window will be shown. Press OK to disconnect all participants.
3. After disconnecting the participants, the End Conference button is enabled.

Disconnecting all participants will not end the conference itself; this will just disconnect all the participants in the conference. The conference can be restarted by pressing the green Redial Action button.
NOTE: Disconnecting all participants from an Ad-hoc conference will terminate the conference.

End Conference

To end the conference completely:

1. Press the End Conference button. This button is only available when there are no active calls left in the conference.
2. A confirmation window will be shown. Press OK to end the conference.
3. The conference will loose its configuration and will be set to [idle] in the MCU Overview page.

Participants

Shows the name of the participants. You may change the displayed name in the Change view, see Change DIsplay Name for details. You may also reset the participant name to the original system name.

Status

Shows the status of the connection:

Establ Out: Shown during call setup between the participant and the MCU.

Alerting: Waiting for the participant to answer the outgoing call.

Connected, 384 kbps: The participant is connected at 384 kbps bandwidth.

Requesting Password: To join the conference. the participant is requested to enter the conference password.

Initiating Encryption: Encryption is being initiated between the MCU and the participant.

Disconnected: The participant has either disconnected or been disconnected by the conference administrator.

Clear Out: The MCU is currently disconnecting the participant.

Net

Shows the network protocol used for the connection:

H.323: The participant is connected on IP using the H.323 protocol.

H.320: The participant is connected on ISDN or V.35 using the H.320 protocol.

SIP: The participant is connected on IP using the SIP protocol

ISDN: The participant is connected on ISDN as a telephone.

Audio

Shows the audio status for each participant (not supported for SIP calls):


The participant is transmitting audio.


The participant has muted the microphone.

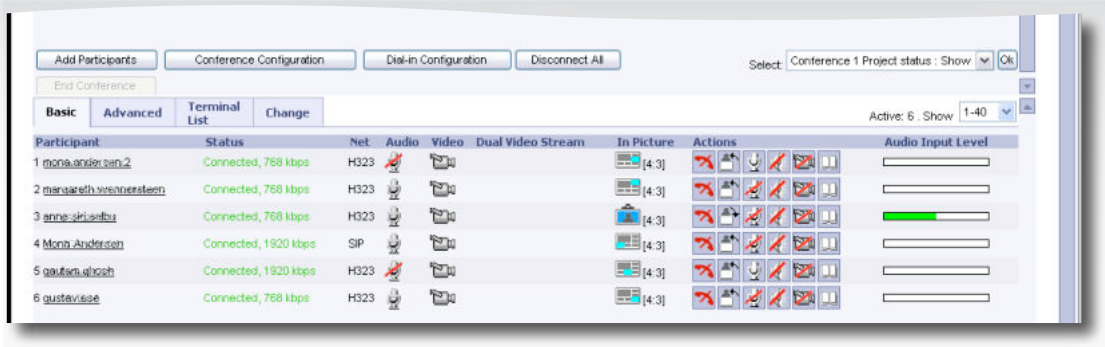

The participant is connected as a telephone.


The conference administrator has manually muted the participant.

Overview MCU, *cont...*

Conference status, the Basic tab

The Conference Status page shows information about a conference and lets you control the conference.



Basic View - Provides a basic overview of all the participants in the conference.

Advanced View - Gives detailed information on the audio in, audio out and video in protocols and the bandwidth used by these protocols.

Terminal List - List the participants in the conference. If the conference is cascaded to another conference, this will be shown here.

Change - The change view is used to modify endpoint constraints or move the endpoint to another conference.

Video

Shows the video status for each participant.

The participant is transmitting video.

The participant is not transmitting video, for instance if the participant has set the video to Off or if the conference administrator has manually muted the participant.

Dual Video Stream (DuoVideo^{TF}/H.239/BFCP)

Shows the Dual Video Stream status for each participant.

The participant is transmitting Dual Video Stream.

The participant is receiving Dual Video Stream.

In Picture

Shows the current Conference Layout and where each participant is displayed in the outgoing video image from the MCU. Some examples:

The participant is in a Voice Switched conference and currently transmitting video to all the other participants.

The participant is in a conference with 5+1 Split layout and is currently transmitting video to all the other participants in the larger quadrant.

The participant is in a conference with 7+1 Split layout and is currently transmitting video to all the other participants in the larger quadrant.

The participant is in a conference with 4 Split layout and is currently transmitting video to all the other participants in the top left quadrant.

The participant is in a conference with 9 Split layout and is currently transmitting video to all the other participants in the top left quadrant.

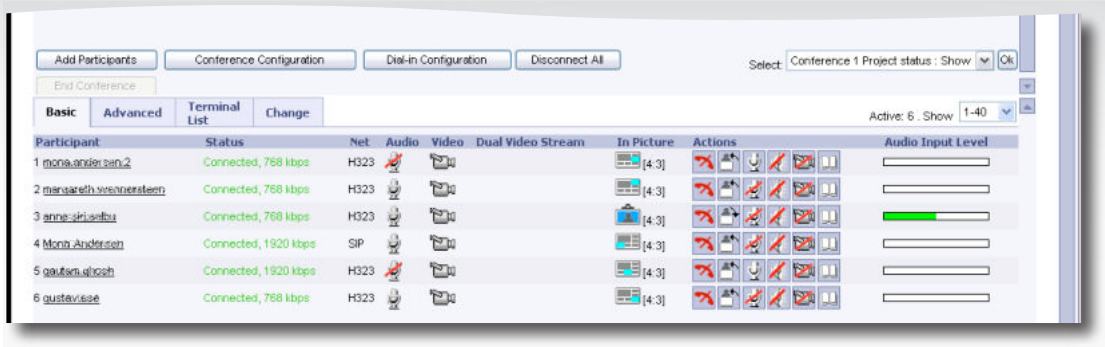
The participant is in a conference with 16 Split layout and is currently transmitting video to all the other participants in the top left quadrant.

The participant has requested the floor or has been assigned the floor by the conference administrator and is transmitting video to all the other participants.

Overview MCU, *cont...*

Conference status, the Basic tab

The Conference Status page shows information about a conference and lets you control the conference.



Basic View - Provides a basic overview of all the participants in the conference.

Advanced View - Gives detailed information on the audio in, audio out and video in protocols and the bandwidth used by these protocols.

Terminal List - List the participants in the conference. If the conference is cascaded to another conference, this will be shown here.

Change - The change view is used to modify endpoint constraints or move the endpoint to another conference.

Actions

During the conference, the conference administrator is able to control each participant.

To disconnect a participant, press the Disconnect button. A confirmation window will be shown. Press OK to disconnect.

If a participant has been disconnected from the conference, the Redial button is shown. Press the button to reconnect the participant. The button is only available for outgoing MCU calls.

To assign the floor to a participant, press the Assign Floor button. The participant is then transmitting video to all the other participants. The participants can also perform this function themselves, if they support the Request Floor (H.243 MVC) functionality.

To release the floor again, press the Release Floor button. This button is shown if the floor has been assigned by the conference administrator or if the participants have requested the floor themselves.

To mute the microphone of a participant, press the Mute Microphone button and the participant will not be heard by the other participants. Note that muted participants will not be able to un-mute themselves, since this was done from the MCU. The participants can, independently of this function, mute their microphone locally on their system.

To un-mute the microphone of a participant again, press the Un-Mute Microphone button. This button is only shown if the audio has been muted by the conference administrator.

To mute the loadspeaker of a participant, press the Mute Loudspeaker button and the participant will not receive any audio from the conference. Note that muted participants will not be able to un-mute themselves, since this was done from the MCU.

To un-mute the loadspeaker of a participant again, press the Un-Mute Loadspeaker button. This button is only shown if the audio has been muted by the conference administrator.

To mute the video of a participant, press the Mute Video Site button and the participant will not be seen by the other participants. Note that muted participants will not be able to un-mute themselves, since this was done from the MCU.

To un-mute the video of a participant again, press the Un-Mute Video Site button. This button is only shown if the video has been muted by the conference administrator.

To add a participant to the TANDBERG MPS Phone Book press the Phone Book button.

Audio Input Level

Shows the measured input level from each of the conference participants.

Overview MCU, *cont...*

Conference status, the Advanced tab

The Conference Status page shows information about a conference and lets you control the conference.

Basic View - Provides a basic overview of all the participants in the conference.

Advanced View - Gives detailed information on the audio in, audio out and video in protocols and the bandwidth used by these protocols.

Terminal List - List the participants in the conference. If the conference is cascaded to another conference, this will be shown here.

Change - The change view is used to modify endpoint constraints or move the endpoint to another conference.

The Advanced View

The Advanced view gives a more detailed picture of the conference than the Basic view.

Audio In/Out: Shows the audio protocol in use, and bandwidth used by the protocol.

Video In: Shows the video protocol in use, and bandwidth used by the protocol.

Participant Info Page

To see all details about the call for each participant, click in the participant name and Participant information page is shown.

Call Info

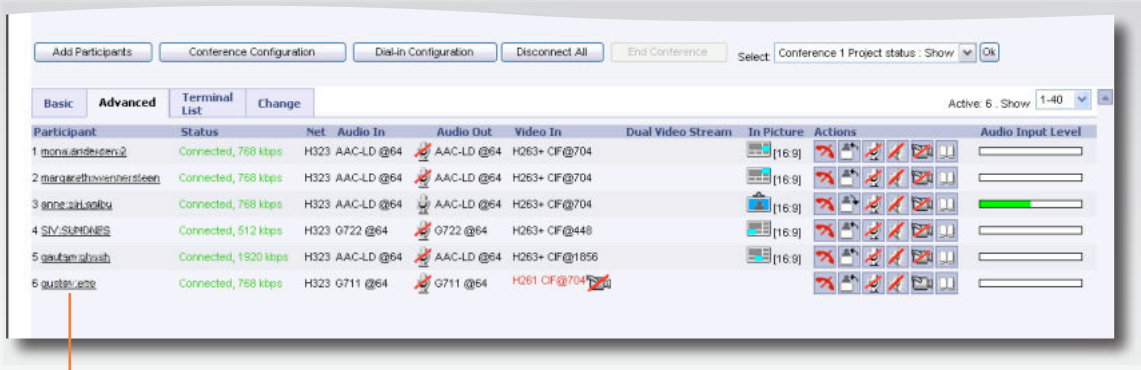
Contains information about Call Status, Call, In/Out/Q.931 Rate, Number, Dial In Number, TerminalID, TerminalName, Duration and Encryption.

Media Info

Contains information about Audio with Dynamic Rate, Packet Loss, Packet Drop, Jitter, RSVP, Local Address and Remote Address; and Video with Dynamic Rate, Packet Loss, Packet Drop, Jitter, RSVP, Local Address and Remote Address; and Dual Stream, Data and Cap Set.

Refresh: Press the Refresh button to do a quick refresh of the page.

Previous/Next: Press the Previous or Next button to see the previous or next participant page.



Click on a participant name to show the Call Info page.

Overview MCU, *cont...*

Conference status, the Terminal List tab

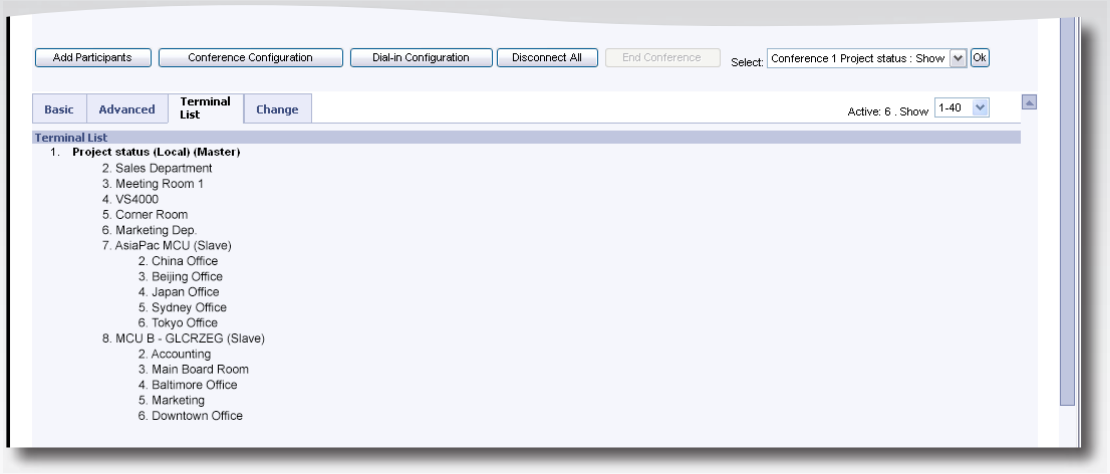
The Conference Status page shows information about a conference and lets you control the conference.

Basic View - Provides a basic overview of all the participants in the conference.

Advanced View - Gives detailed information on the audio in, audio out and video in protocols and the bandwidth used by these protocols.

Terminal List - List the participants in the conference. If the conference is cascaded to another conference, this will be shown here.

Change - The change view is used to modify endpoint constraints or move the endpoint to another conference.



Terminal List

The Terminal List will list all participants in conferences on the MCU.

MCUs in Cascade

The TANDBERG MPS supports two-level cascading, so the TANDBERG MPS can be cascaded with other TANDBERG MPSs to increase the number of participants in one conference.

How to Initiate Cascaded Conference

By simply dialling from one TANDBERG MPS to other TANDBERG MPSs one will achieve a distributed setup.

Master/Slave

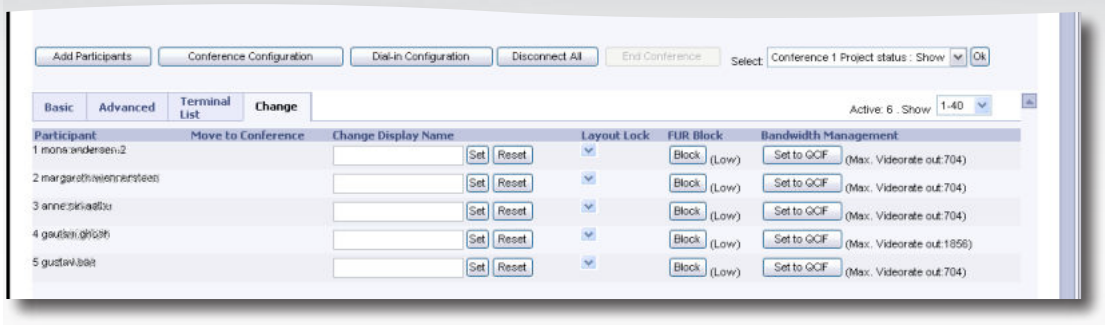
The TANDBERG MPS dialling out will be defined as the Master MCU. All other MCUs will be defined as Slave MCUs.

Read about cascaded MCUs in the Distributed MCUs chapter in the Appendices section.

Overview MCU, *cont...*

Conference status, the Change tab

The Conference Status page shows information about a conference and lets you control the conference.



Basic View - Provides a basic overview of all the participants in the conference.

Advanced View - Gives detailed information on the audio in, audio out and video in protocols and the bandwidth used by these protocols.

Terminal List - List the participants in the conference. If the conference is cascaded to another conference, this will be shown here.

Change - The change view is used to modify endpoint constraints or move the endpoint to another conference.

Participant

Shows the participant number or system name.

Applies for H.320, H.323 and SIP audio/video participants.

Move to Conference

Move the participant to another conference (both created conferences and ongoing conferences).

- Press the Move button to move the participant to the other conference.
- The conference must be started in advance and should have the same basic settings.
- It is not possible to move a H.323 call to an encrypted conference.

Change Display Name

Change the participant name/text which is displayed in the bottom of the Conference Snapshots window. This will replace the system name if available.

- Press the Set button to change the displayed name.
- Press the Reset button to restore the original system name.

The Participant Identifier must be set to On or Auto to show the name of a participant in the Conference Snapshot window.

Layout Lock

Lock the participant to a fixed position in the conference view. This could be used when you want one particular participant to be displayed in the big square in a split Conference Layout.

The layout in use will decide how this will look like.

Please refer to Conference Layout > Lecture Mode for details.

Bandwidth Management

Press the Set to QCIF button to force the endpoint (video system) to move to a low rate encoder.

This is to enable more bandwidth to the other participants in the conference.

It is not possible to force the endpoint back to a higher-rate encoder without reconnecting the call.

FUR Block

The conference administrator can block FUR (Fast Update Request) for a participant. This will prevent Fast Update Requests for the participant.

- Press the Block button in the FUR Block column to block fast update request for the selected participant.

FUR from an endpoint (video system) may degrade the experience for other participants. Selecting FUR Block to On may increase the quality for the other participants.

To block FURs from all sites, set Network Error Handling to FURBlock and FUR Block Sites to On in the Conference Configuration page.

FUR Rate

The system will display the FUR Rate; High or Low, for each participant.

Maximum Video Rate Out

Shows the maximum video rate, reported from the endpoint (video system), which a site can receive.

The Gateway

Gateway Features

Encryption (Secure conference[TF])

Secure conference[TF] AES (128 bit) and Secure conference[TF] DES (56 bit) are both supported through the Gateway.

Downspeeding[TF]

When requested quality (bandwidth) cannot be established, Downspeeding[TF] makes the connection establish on as high quality as possible. If channels are dropped during a video conference, Downspeeding[TF] will ensure the connection being automatically maintained without interruption.

Dual Video Streams

DuoVideo[TF]/H.239 are supported from both ISDN and IP and allows participants at the far end to simultaneously view a presenter and a live presentation.

Digital Clarity[TF]

Participants enjoy presentations of exceptionally high quality resolution video.

Natural Video[TF]

60 fields per second true interlaced picture (iCIF).

H.264 support through Gateway[TF]

Superb video quality supporting the ITU video standard H.264.

AAC-LD support through Gateway[TF]

True standards-based CD-quality audio supporting the audio standard AAC-LD.

Full H.243 Transparency

H.243 Multipoint Control Unit (MCU) Transparency allows seamless MCU control through the Gateway.

Far End Camera Control using H.224 (H.281)

Far End Camera Control using H.224 (H.281) allows seamless Far End Camera control through the Gateway.

Intelligent Packet Loss Recovery (IPLR)

If the Gateway experiences packet loss from an IP endpoint, it will ask the endpoint to handle packet loss. This requires Intelligent Packet Loss Recovery (IPLR) functionality on the video system (endpoint).

Text Chat (T.140)

Text Chat (T.140) is supported through the Gateway.

Gateway Support for OD/HD

- H.264 w288p, 400p, w448p, w576p
- H.263 w288p, 400p, w488p

Gateway Capacity

Each Media Board can support up to 7680 kbps of total bandwidth.

Each Gateway call consists of two connections. Example: A 384 kbps Gateway call has one H.323 site connected at 384 kbps and one H.320 site connected at 384 kbps. This means that this call is utilizing 768 kbps worth of bandwidth (384k + 384k = 768k).

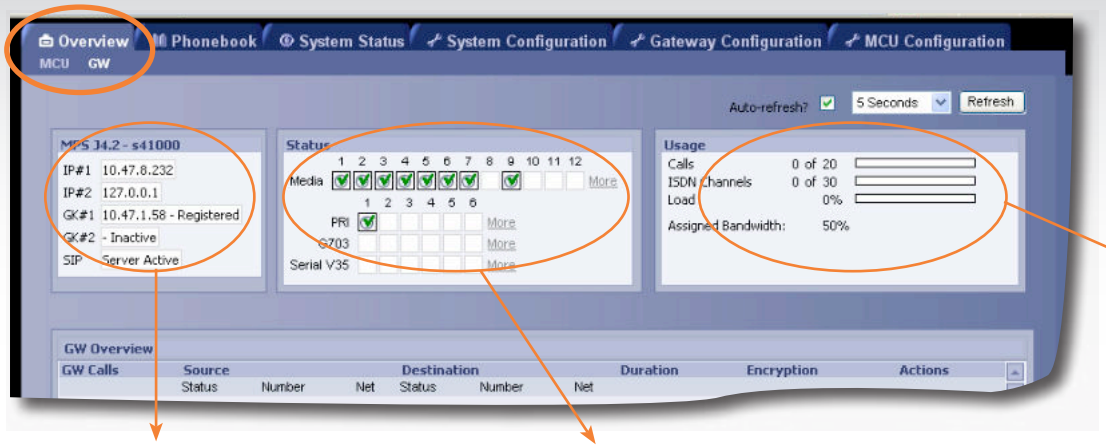
Each Gateway option on the MPS allows for a maximum of 10 concurrent calls. This means that each Gateway option can support up to 10 concurrent calls at 384 kbps.

- Decreasing the bandwidth of each call will not increase the maximum number of calls possible.
- Increasing the bandwidth of each call will decrease the maximum number of calls.

Below is a chart outlining the maximum number of calls possible for each bandwidth that is supported on the Gateway.

Bandwidth in Use on Gateway	Maximum Number of Calls	Remaining Bandwidth
Telephone 64 kbps	Max 10 calls	
64 kbps	Max 10 calls	
128 kbps	Max 10 calls	
192 kbps	Max 10 calls	
256 kbps	Max 10 calls	
384 kbps	Max 10 calls	
512 kbps	Max 7 calls	This would be 7168 kbps total which leaves 512 kbps remaining for lower rate calls
768 kbps	Max 5 calls	
1152 kbps	Max 3 calls	This would be 6912 kbps total which leaves 768 kbps remaining for lower rate calls
1472 kbps	Max 2 calls	This would be 5888 kbps total which leaves 1792 kbps remaining for lower rate calls
1536 kbps	Max 2 calls	This would be 6144 kbps total which leaves 1536 kbps remaining for lower rate calls
1920 kbps	Max 2 calls	

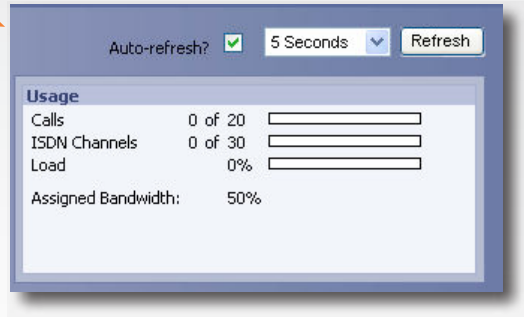
Gateway Usage Information



The Usage pane for Gateway calls shows each active call through the Gateway (Load, ISDN channels and number of calls).

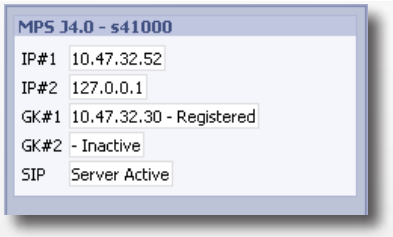
Right above the Usage pane you can:

- Set Auto-Refresh to On/Off.
- Adjust the Refresh Rate (2-30 seconds)
- For a quick refresh click the Refresh button.



The General field displays some basic information:

- Software version
- IP address(es)
- Gatekeeper address and status
- SIP status



The Status field displays the connections status for the installed boards and cards:

- Media boards
- ISDN PRI card
- ISDN G.703 card
- SERIAL card

Indicates OK

Indicates a warning

Indicates an error

Click More to see more details.

The status of the Gateway resources:

- Calls 0 of 10: Indicates that 0 calls are connected through the Gateway. The total number of supported calls in this configuration is 10.
- ISDN Channels 0 of 45: Indicates that 0 ISDN channels are used on the TANDBERG MPS. The total number of supported ISDN channels in this configuration is 45.
- Load 0%: Indicates 0% load on the Gateway resources.
- Assigned Bandwidth 50%: Indicates that 50% of the bandwidth is assigned for Gateway calls.
- Load Limit 100%: Indicates that the Load Limit is set to 100%. When the Resource Usage reaches the “Busy on Load”-limit, the Gateway will signal this to the Gatekeeper. The Gatekeeper will then try to route calls through other Gateways. This is done to maintain availability for incoming ISDN calls when using multiple Gateways.

Gateway Calls Overview

GW Calls Shows each active call through the gateway.

- [Idle] No call is active.
- Active Call A call is active. Click on Active Call to see call info in details.

Source Shows the Status of the incoming call to the Gateway. Shows the Number and which Network the incoming call is using.

Destination Shows the status of the outgoing call from the gateway, the number and which network the call is using.

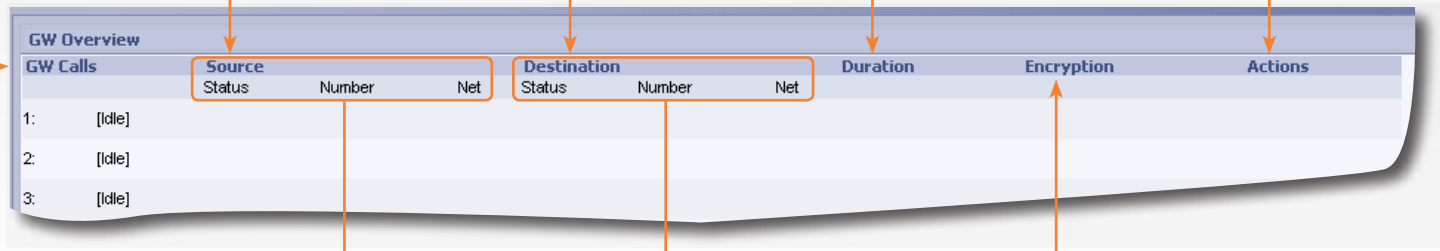
Duration Shows the call duration of the current call.

Actions

When a call is active, the Administrator or Operator can disconnect the call, or transfer the call from one IP endpoint to another.

Disconnect Gateway Call Will disconnect the selected gateway call. A confirmation window will be shown.

Call Transfer The H.323 Call Transfer window is displayed. The Administrator or Operator can select an IP entry from the Phone Book or enter a H.323 ID/IP address and transfer the H.323 (IP) call to another IP endpoint. If the transfer fails, the call will be disconnected.



Status

Shows the current status of the call.

- Idle No active call.
- Establ Out Gateway is calling out to destination.
- Alerting Call is being connected.
- Connected Call is connected.
- Clear out Call is being disconnected.

Number ISDN or IP number.

Network

- ISDN Call connected is using the H.320 protocol over ISDN.
- IP/H.323 Call connected is using the H.323 protocol over IP.

Encryption (Secure Conference[TF])

The Gateway supports Secure Conference DES and AES.

Indicates that DES (56 bit) encryption is used.

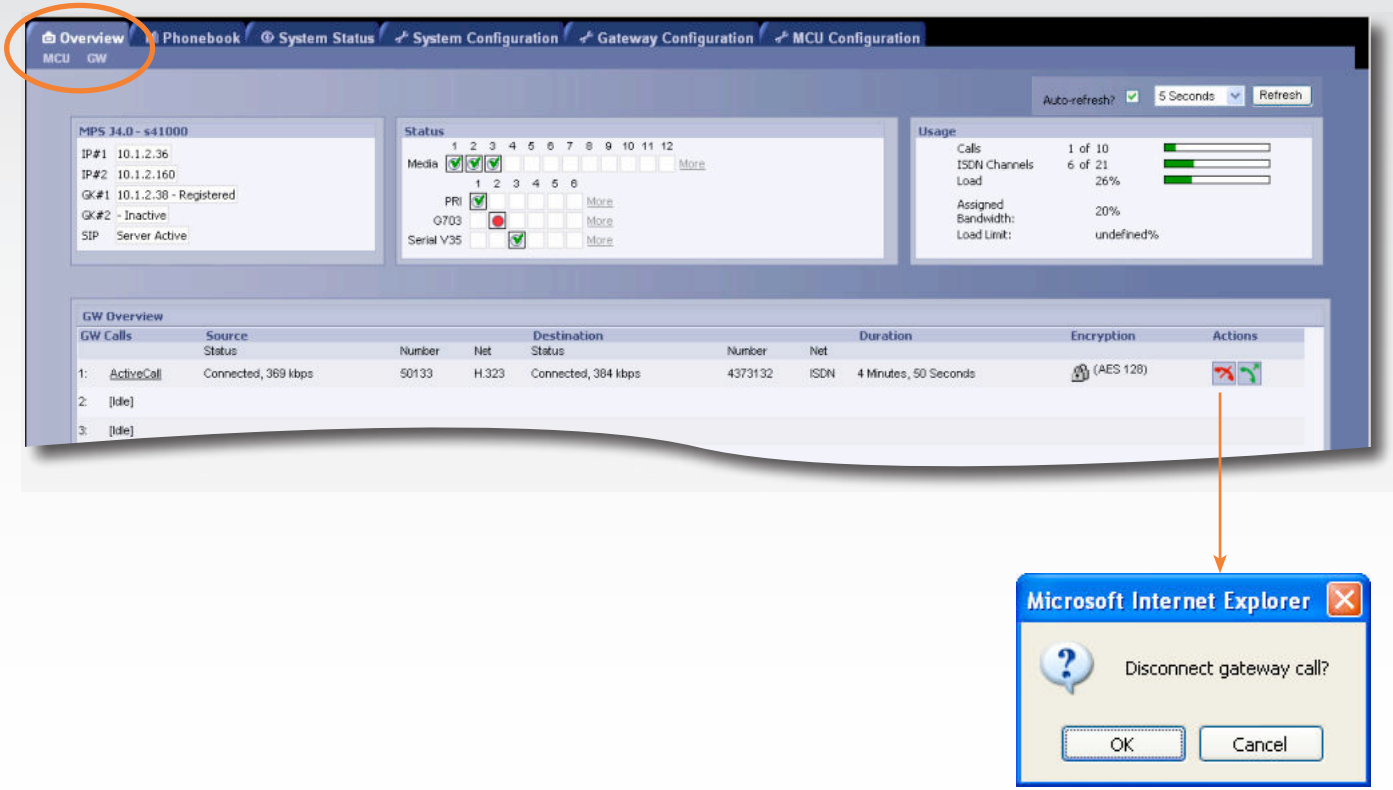
Indicates that AES (128 bit) encryption is used.

Indicates that the call is not encrypted.

Disconnect a Gateway call

Actions - When a call is active the Administrator or Operator can Disconnect the call, or Transfer the call from one IP endpoint to another.

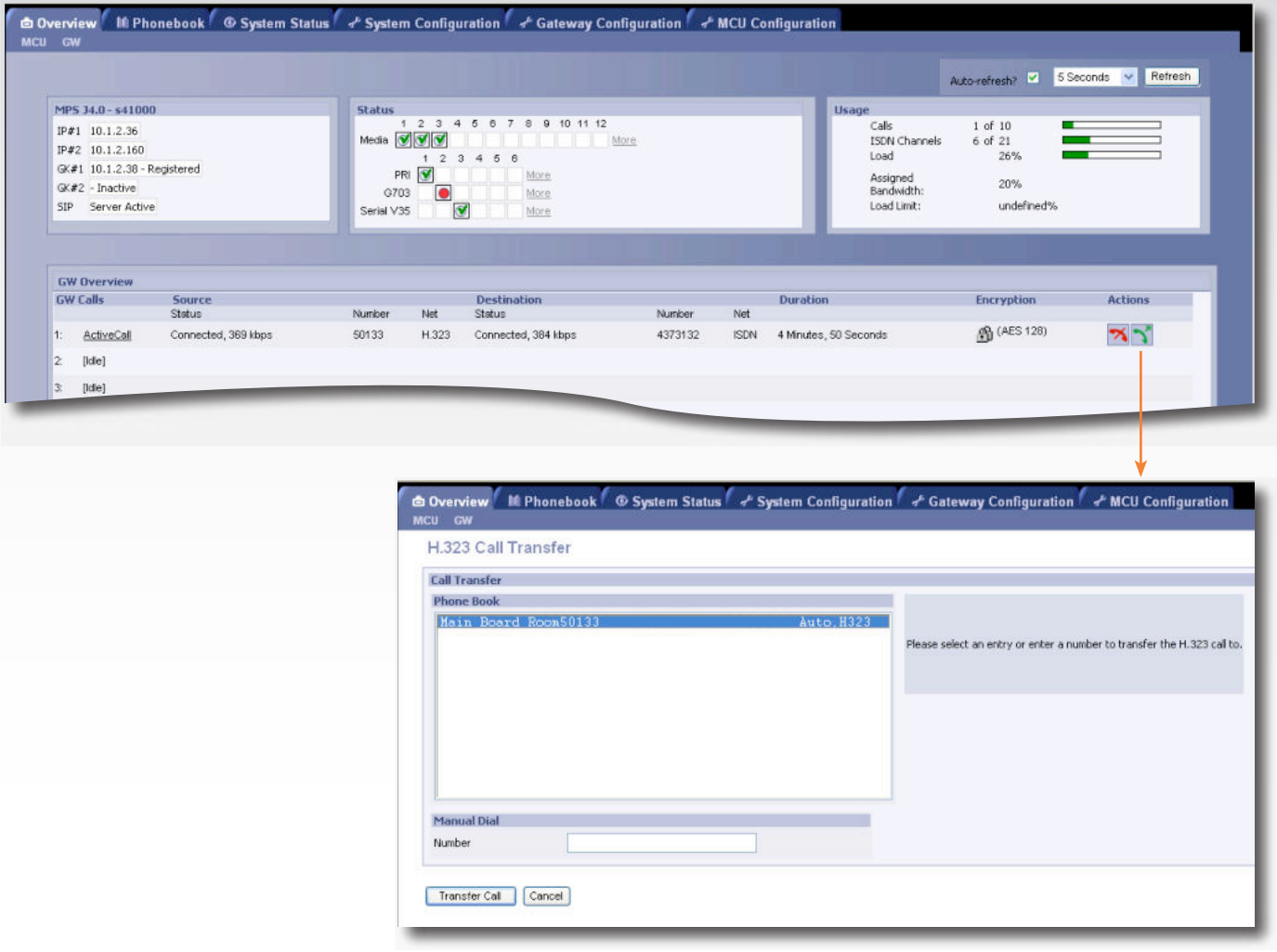
Select the Overview tab and GW



1. Identify which call to be disconnected from the Gateway Overview page.
2. Press the Disconnect Call actions button to disconnect the selected Gateway call.
3. A confirmation window will be shown:
   - Press OK to Disconnect the call
   - Press Cancel to discard the action

Transfer a Gateway call

Actions - When a call is active the Administrator or Operator can Disconnect the call, or Transfer the call from one IP endpoint to another.

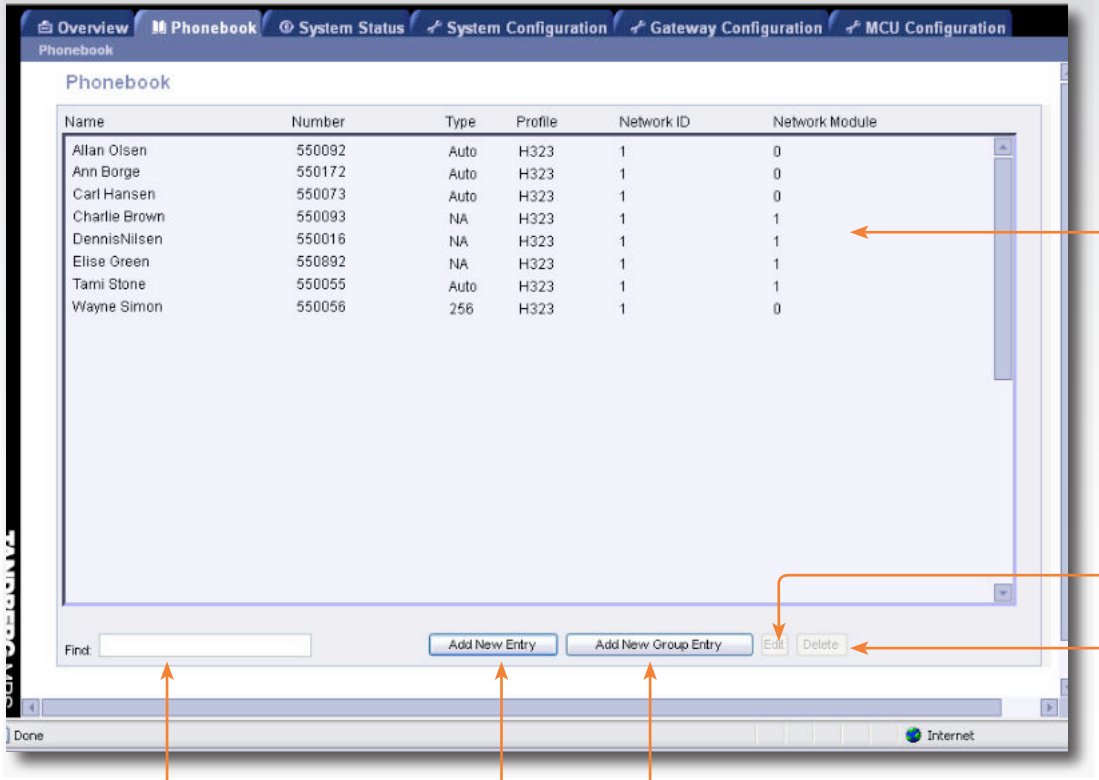


1. Identify which call to transfer from the Gateway Overview page.
2. Press the  Transfer Call actions button.
3. A H.323 Call Transfer window will be shown.
4. Select a H.323 entry from the Phone Book or enter an IP address or E.164 alias into the Manual Dial field.
   - Press Transfer Call button to initiate the transfer
   - Press the Cancel button to discard the transfer.

If the transfer fails the call will be disconnected.

The Phone book

Select the Phonebook tab



All settings for single entries and group entries are explained on the following pages.

Edit: Select an entry from the list and press the Edit button to open the Edit Entry dialog box.

Delete: Select an entry from the list and press the Delete button to delete an entry.

A confirmation dialog will show to let you confirm the action.

Number of Entries

There is support for:

- Up to 250 local entries in the phone book.
- Up to 16 Group Entries.
- Each Group Entry can have up to 32 members.

Search

Enter the first characters of the name or the complete name to find a name in the list.

Add New Entry

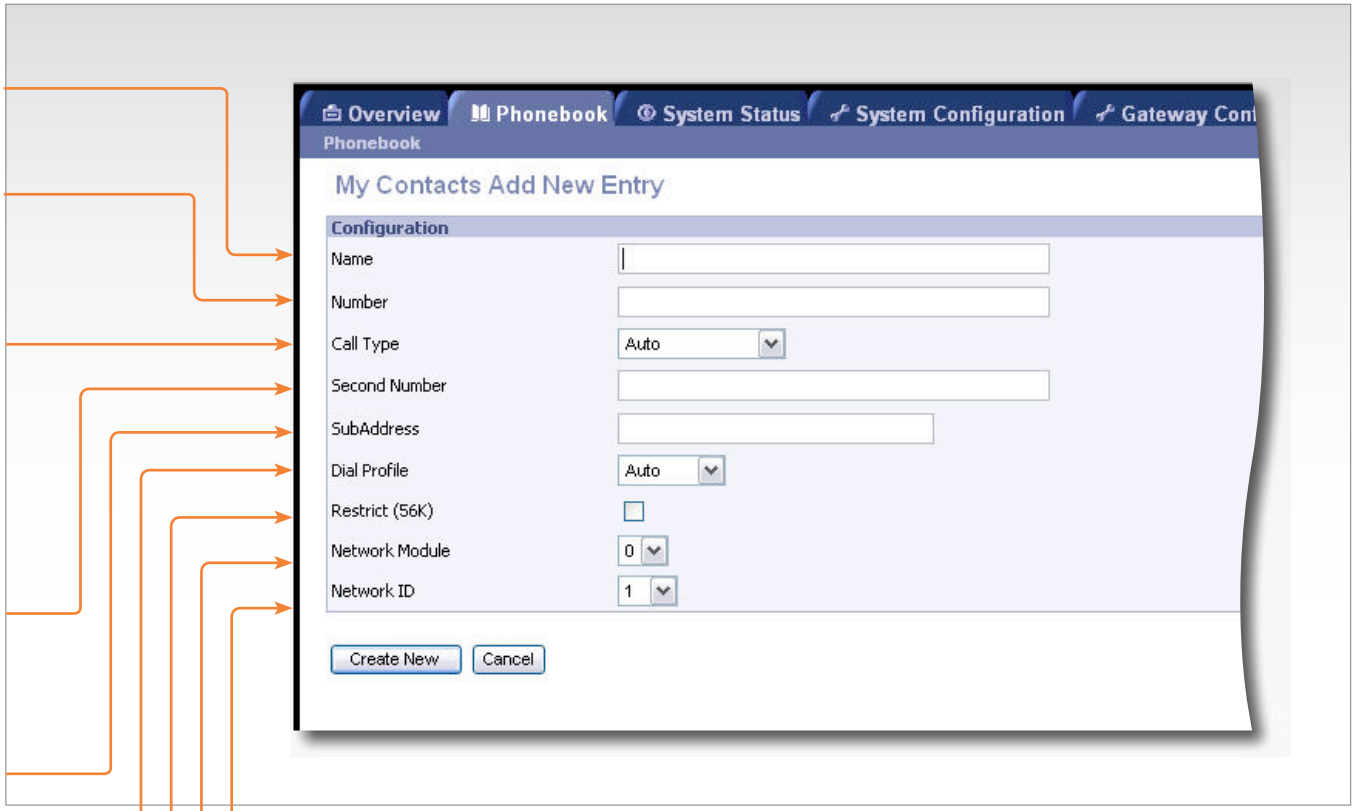
Press the Add New Entry button to open the Add Entry dialog box (*discussed on the following pages*).

Add New Group Entry

Press the Add New Group Entry button to open the Add Group Entry dialog box (*discussed on the following pages*).

The Phone book, *cont...*

My Contacts, Add New Entry



Name
Enter the name of the phone book entry.

Number
Enter the video number, telephone number or RS366 dial string.

For G.703 Leased Line calls, this is the call number (1 to 5).

Call Type
Telephone: Set to Telephone if the number entered above is a telephone number.

Auto: When set to Auto (default) the configured Maximum Conference Rate will be used.

Custom Selection: Select the call rate to be used.
(64 kbps, 2x64 (H.221) kbps, 128 kbps, 192 kbps, 256 kbps, 320 kbps, 320 kbps, 384 kbps, H0 (384 kbps), 512 kbps, 768 kbps, 1152 kbps, 1472 kbps, 1920 kbps)

Second Number
If two numbers are required, both should be specified for 2x64 kbps and 2x56 kbps calls.

Leave blank (default) if the endpoint only has one number.

Subaddress
The Subaddress is used to address different participants on the same ISDN line or TCS4 dialling.

Dial Profile
Select which dial profile to use for this phone book entry.

Auto: The MPS will automatically select the appropriate profile

ISDN: ISDN will always be used for dialling.

H.323: H.323 will always be used when dialling.

SIP: SIP will always be used when dialling.

If defined, custom Dial Profiles (Network Profiles) will show in the list.

Restrict (56K)
Select Restrict (56K) to use 56 kbps per ISDN B-channel.

Network ID
Used to identify port or interface number within a network module. Enter a value between 1 and 32:

- Specify which IP network to use, only 1 and 2 are valid values (optional).
- Specify which V.35 port to use (mandatory).

Create New
Select Create New to save the new Phone Book Entry.

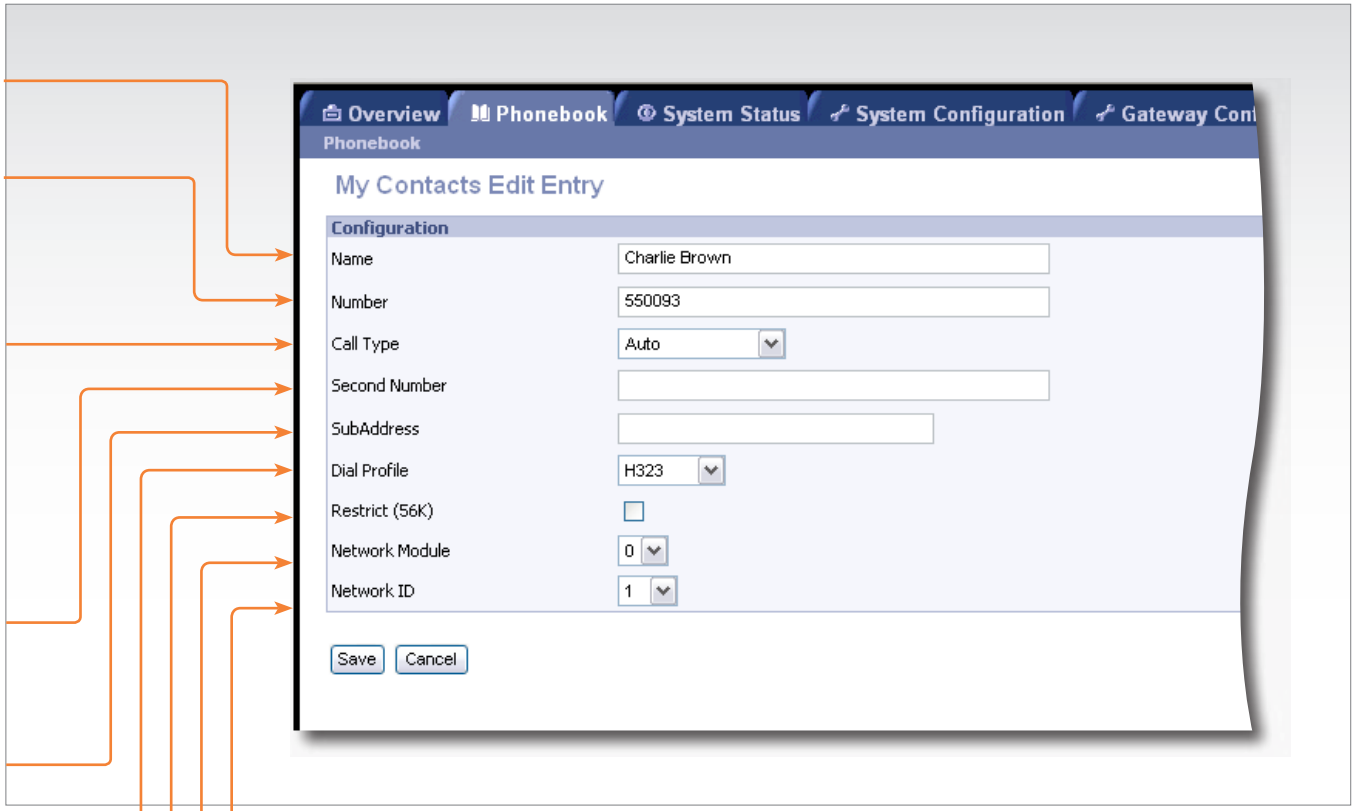
Network Module
Specify which Network Interface Card to use for outgoing calls. Enter a value between 0 and 6.

- Optional for ISDN calls
- Mandatory for V.35 calls and G.703 Leased Line calls.

Cancel
Select Cancel to discard all changes and return to the Phone Book.

The Phone book, *cont...*

My Contacts, Edit New Entry



Name
Enter the name of the phone book entry.

Number
Enter the video number, telephone number or RS366 dial string.

For G.703 Leased Line calls, this is the call number (1 to 5).

Call Type
Telephone: Set to Telephone if the number entered above is a telephone number.

Auto: When set to Auto (default) the configured Maximum Conference Rate will be used.

Custom Selection: Select the call rate to be used.
(64 kbps, 2x64 (H.221) kbps, 128 kbps, 192 kbps, 256 kbps, 320 kbps, 320 kbps, 384 kbps, H0 (384 kbps), 512 kbps, 768 kbps, 1152 kbps, 1472 kbps, 1920 kbps)

Second Number
If two numbers are required, both should be specified for 2x64 kbps and 2x56 kbps calls.

Leave blank (default) if the endpoint only has one number.

Subaddress
The Subaddress is used to address different participants on the same ISDN line or TCS4 dialling.

Dial Profile
Select which dial profile to use for this phone book entry.

Auto: The MPS will automatically select the appropriate profile

ISDN: ISDN will always be used for dialling.

H.323: H.323 will always be used when dialling.

SIP: SIP will always be used when dialling.

If defined, custom Dial Profiles (Network Profiles) will show in the list.

Restrict (56K)
Select Restrict (56K) to use 56 kbps per ISDN B-channel.

Network ID
Used to identify port or interface number within a network module. Enter a value between 1 and 32:

- Specify which IP network to use, only 1 and 2 are valid values (optional).
- Specify which V.35 port to use (mandatory).

Save
Select Save to save the changes to this Phone Book Entry.

Network Module
Specify which Network Interface Card to use for outgoing calls. Enter a value between 0 and 6.

- Optional for ISDN calls
- Mandatory for V.35 calls and G.703 Leased Line calls.

Cancel
Select Cancel to discard all changes and return to the Phone Book.

The Phone book, *cont...*

Add Group Entry

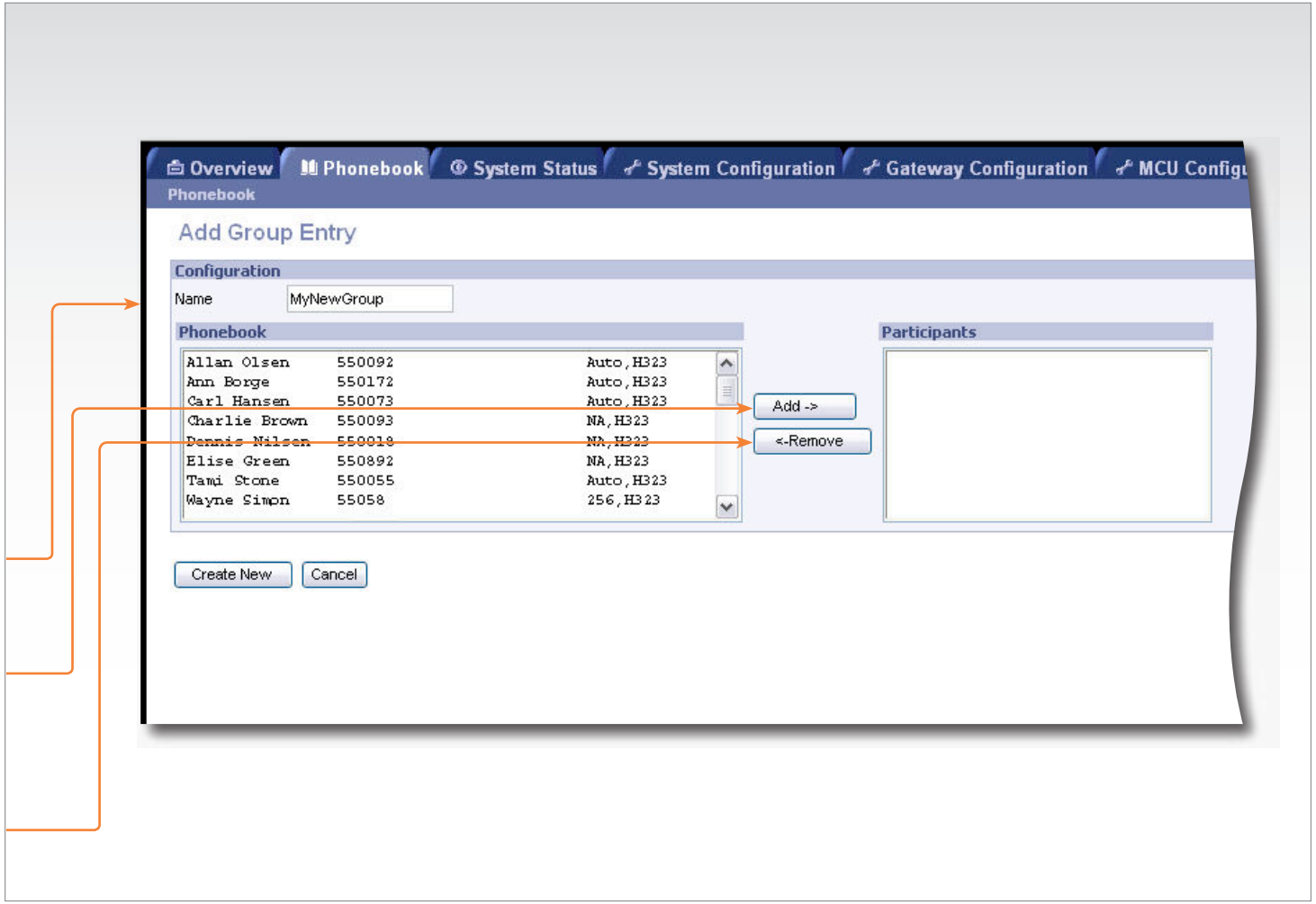
Group Entry

The Group entries are useful for recurring meetings where the same participants meet each time. By doing this, only the group entry has to be selected in the Add Participants menu and the participants are automatically called.

This view is opened from the Phone Book. Select Phone Book from the top menu and press the Add New Group Entry button.

You can define up to 16 Group Entries.

Each Group Entry can have up to 32 members.



Name

Enter a name for the Group Entry.

Add ->

Select a name from the Phone Book list and press the Add -> button to add the name into the Participants list.

<- Remove

Select a name from the Participants list and press the <- Remove button to remove the name from the Participants list.

Create New

Select Create New to save the new Phone Book Group Entry.

Cancel

Select Cancel to discard all changes and return to the Phone Book.

Chapter 4

System status

The System Status gives an overview of the status for the boards and cards installed on the MPS. The information will reflect a fully equipped MPS and may differ from the MPS delivered.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

In this chapter...

H.320 Board Status

H.320 Status

The H.320 Status shows the system clock port, rate, state and type.

If in an error state, there may be problems establishing calls on all H.320 networks (i.e. PRI, G703 and Serial V.35).

Status of H.320 reference clock

The first extension board will always be used as the external clock reference for the H.320 system clock.

PRI/G.703 and V.35 are all synchronous networks which require a clock-reference. The clock reference must be taken from the extension board connected to the first media-blade.

System Clock Port:

The port or interface in use.

System Clock Rate:

The frequency of the clock reference which is between 56 kHz and 1920 kHz. The intervals are in steps of 56 kHz or 64 kHz.

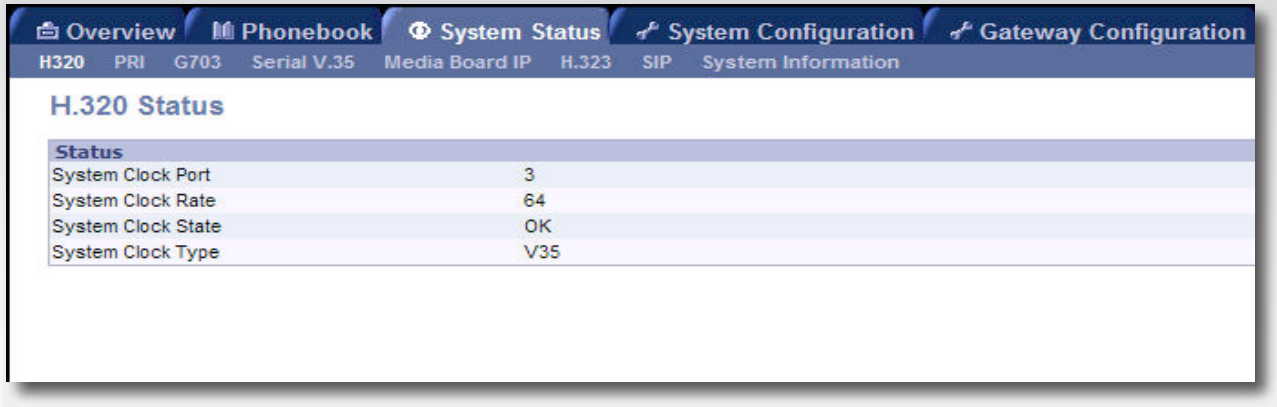
System Clock State:

OK: If OK, the system is using an external clock reference.

Error: If Error, the system has no external reference and clock slips may occur.

System Clock Type:

Which type of extension board in use, V.35 or PRI.



ISDN PRI Board Status

ISDN PRI Board Status

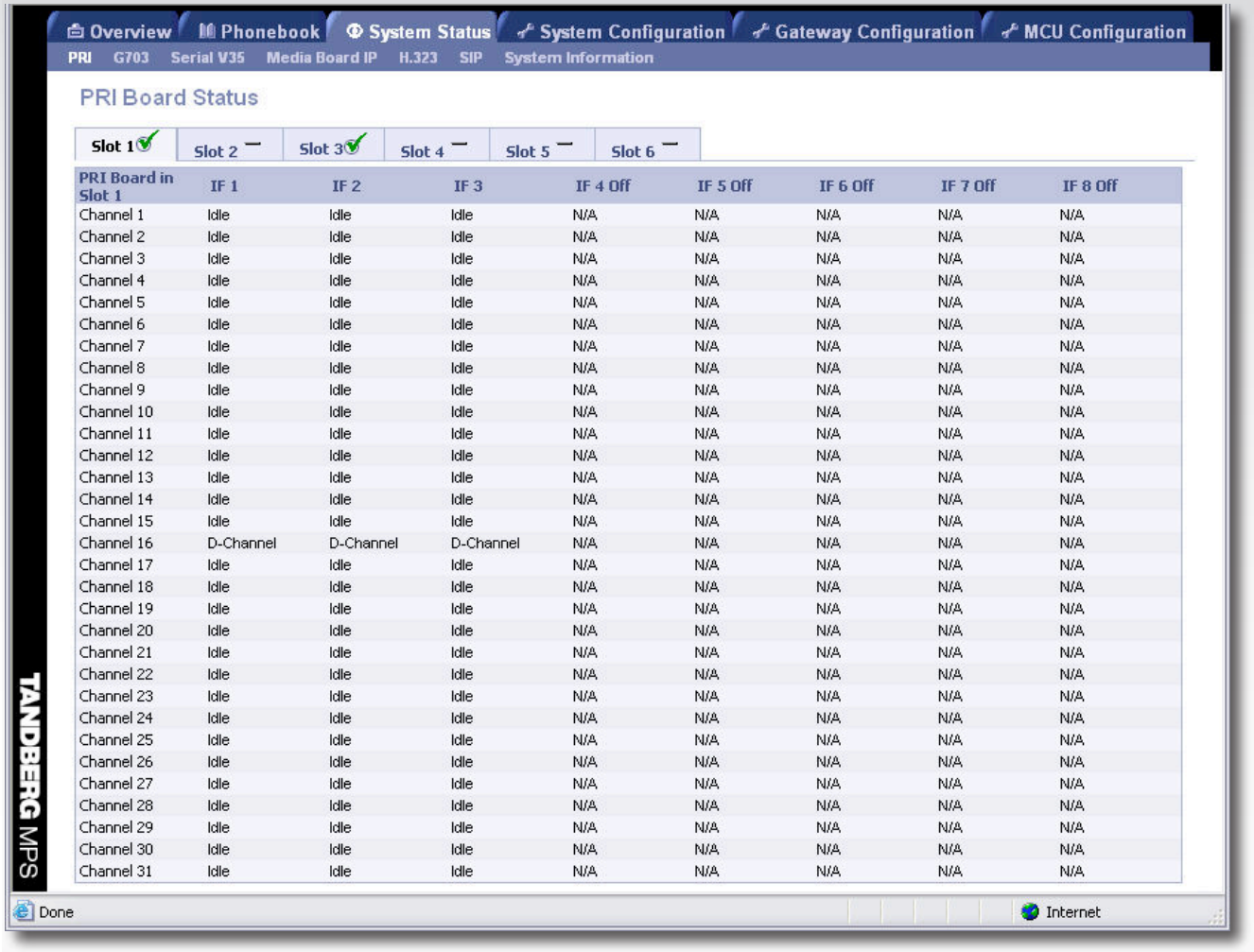
If a PRI cable is disconnected or malfunctioning, or basic configuration is erroneous,

- Red Alarm is displayed in red for the affected interfaces.
- Sync alert is displayed in red for the affected interfaces.
- Clock Slip shows the number of Clock Slips in the G.703 network the last 24 hours. Clock Slips may lead to problems establishing calls and reduced call quality. After correcting the network errors, please reset the counter by pressing the Clear-button.

If a participant has been disconnected, the cause code can be viewed by pressing the link next to the disconnected PRI channel.

The most common cause codes (for ISDN):

1 Unallocated (unassigned) number
2 No route to specified transit network (WAN)
16 Normal call clearing
17 User busy
18 No user responding
21 Call rejected
28 Invalid number format (incomplete number)
29 Facilities rejected
31 Normal, unspecified
34 No Circuit/Channel Available
41 Temporary Failure
58 Bearer Capability Not Presently Available
65 Bearer Capability Not Implemented
69 Requested Facility Not Implemented
81 Invalid Call Reference Value
88 Incompatible Destination
100 Invalid Information Element Contents
102 Recovery On Timer Expiry
127 Interworking, Unspecified

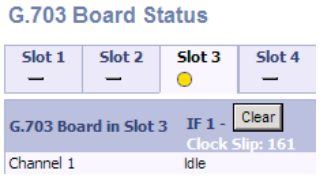


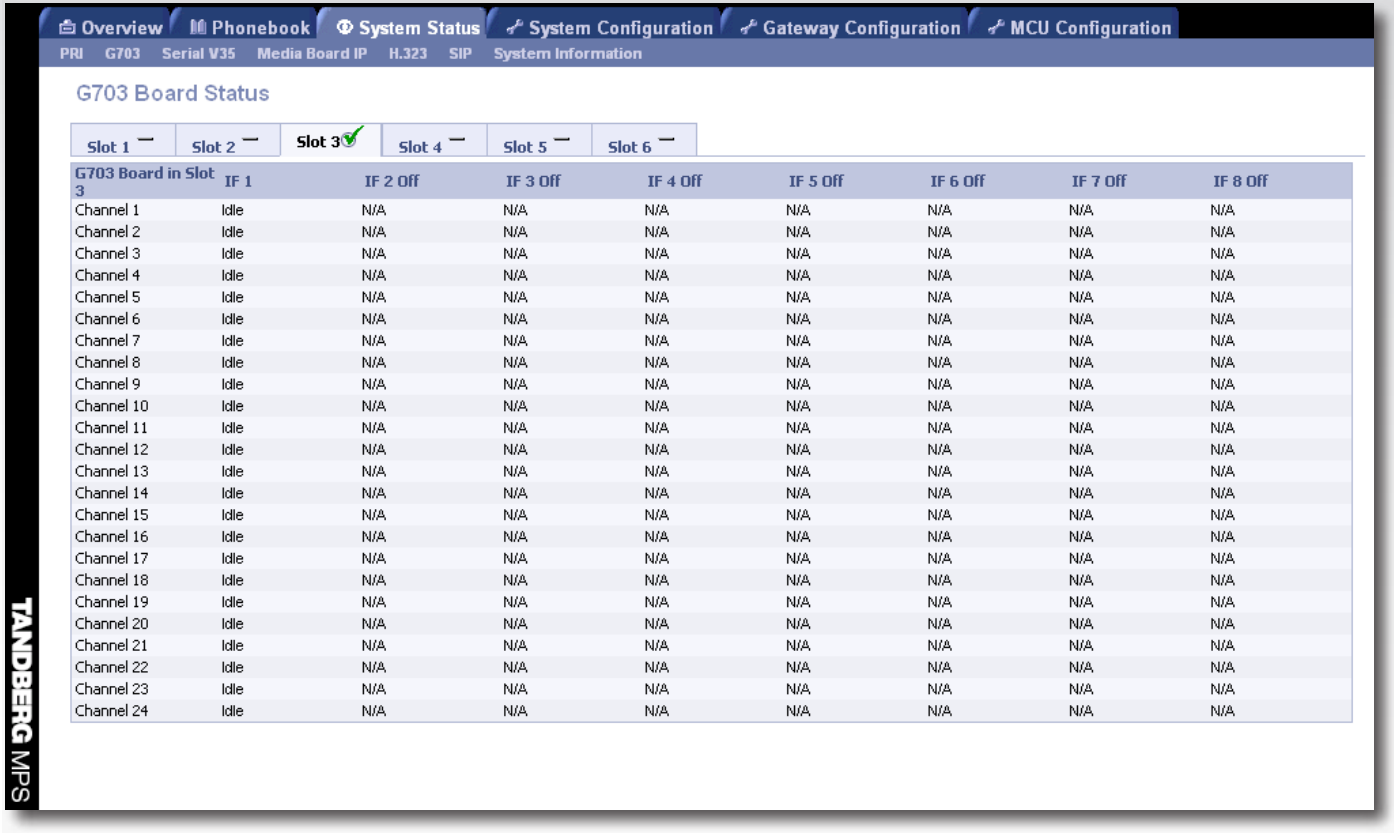
G.703 Board Status

G.703 Board Status

This page gives status information about available channels and which calls are connected and recently disconnected.

- Red Alarm may indicate that the cable is faulty or disconnected.
- Sync alert may indicate that the MPS or your G.703 network is not correctly configured.
- Clock Slip shows the number of Clock Slips in the G.703 network the last 24 hours. Clock Slips may lead to problems establishing calls and reduced call quality. After correcting the network errors, please reset the counter by pressing the Clear-button.





OK Warning Error

Serial V.35 Board Status

Serial V.35 Board Status

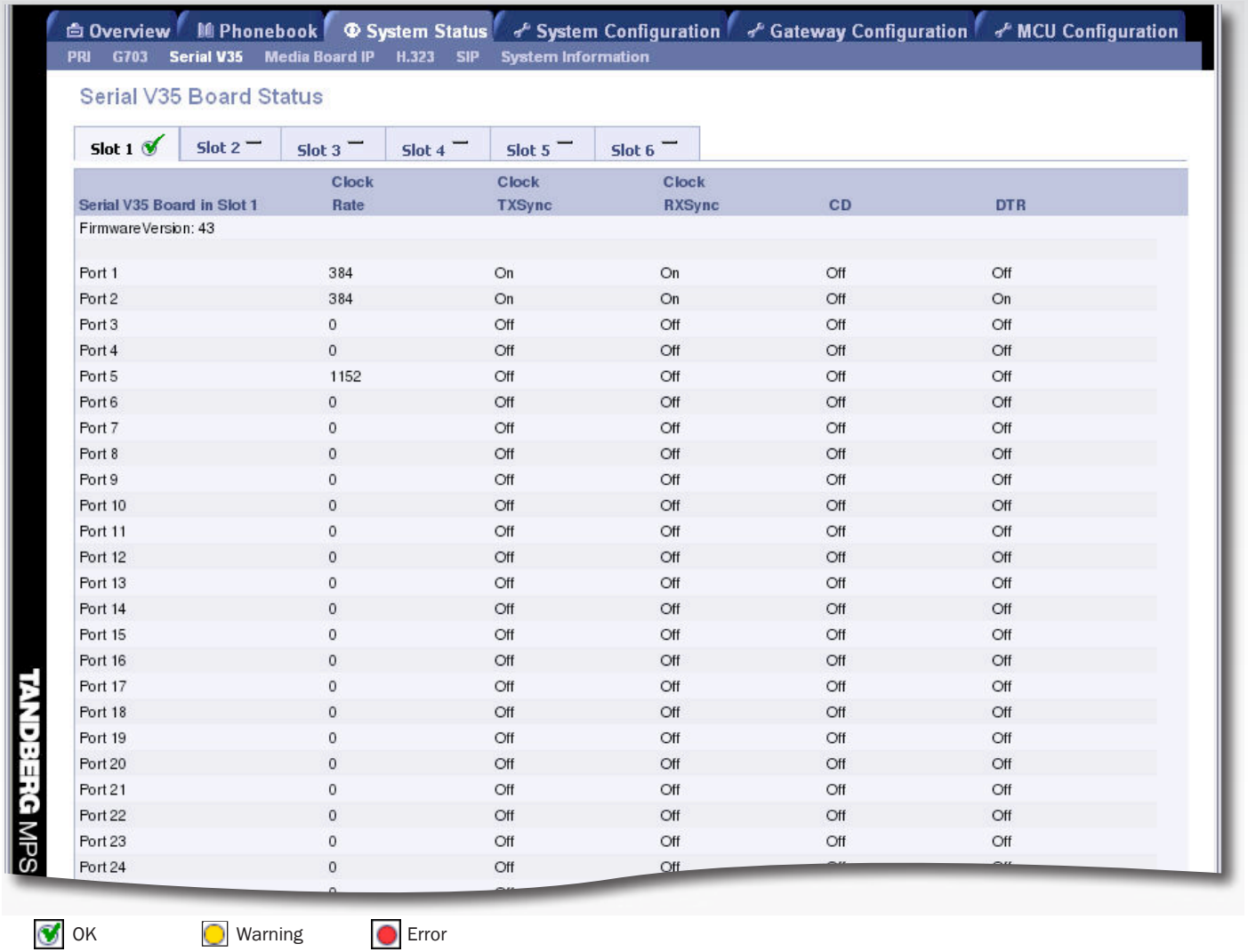
Clock rate: Displays the current clock rate for each individual Serial port. When a port is active in a call this will display the call rate that the port is connected at. If used for external clocking on the MPS this will display the rate of the clock that the MPS is receiving from the network.

Clock TXSync/Clock RXSync: Displays the current Tx data clock and Rx data clock synchronization status for each individual port. If status is On the data clock is synchronized with the internal reference clock of the MPS. If status is Off the data clock is not synchronized with the internal reference clock of the MPS. During a call both RxSync and TxSync should be On. If not there is a problem with the clock synchronization of either the MPS or the external equipment. Lack of clock synchronization may cause severe deterioration of video and audio quality and may even prevent call establishment.

For further information please refer to the System Clock configuration in the System Configuration > Miscellaneous section.

CD: Displays the current status for Carrier Detect for each individual port. When a port is active in a call this will show “On”. When a port is in the idle state this will show “Off”.

DTR: Displays the current status for Data Terminal Ready for each individual port. When a port is active in a call or assigned to a conference this will show “On”. When a port is in the idle state this will show “Off”.



Serial V35 Board in Slot 1	Clock Rate	Clock TXSync	Clock RXSync	CD	DTR
FirmwareVersion: 43					
Port 1	384	On	On	Off	Off
Port 2	384	On	On	Off	On
Port 3	0	Off	Off	Off	Off
Port 4	0	Off	Off	Off	Off
Port 5	1152	Off	Off	Off	Off
Port 6	0	Off	Off	Off	Off
Port 7	0	Off	Off	Off	Off
Port 8	0	Off	Off	Off	Off
Port 9	0	Off	Off	Off	Off
Port 10	0	Off	Off	Off	Off
Port 11	0	Off	Off	Off	Off
Port 12	0	Off	Off	Off	Off
Port 13	0	Off	Off	Off	Off
Port 14	0	Off	Off	Off	Off
Port 15	0	Off	Off	Off	Off
Port 16	0	Off	Off	Off	Off
Port 17	0	Off	Off	Off	Off
Port 18	0	Off	Off	Off	Off
Port 19	0	Off	Off	Off	Off
Port 20	0	Off	Off	Off	Off
Port 21	0	Off	Off	Off	Off
Port 22	0	Off	Off	Off	Off
Port 23	0	Off	Off	Off	Off
Port 24	0	Off	Off	Off	Off

Media Board IP Status

Media Board IP Status

Red alarm indicates the Ethernet interface on the Media Board is down.

IP Address

Shows the IP Address of the Media Processing Board.

IP Subnet Mask

Shows the IP Subnet Mask of the Media Processing Board.

IP Gateway

Shows the IP Gateway of the Media Processing Board.

MAC Address

Shows the MAC address of the Media Processing Board.

Ethernet Speed

Shows the speed on the Ethernet interface of the Media Processing Board.

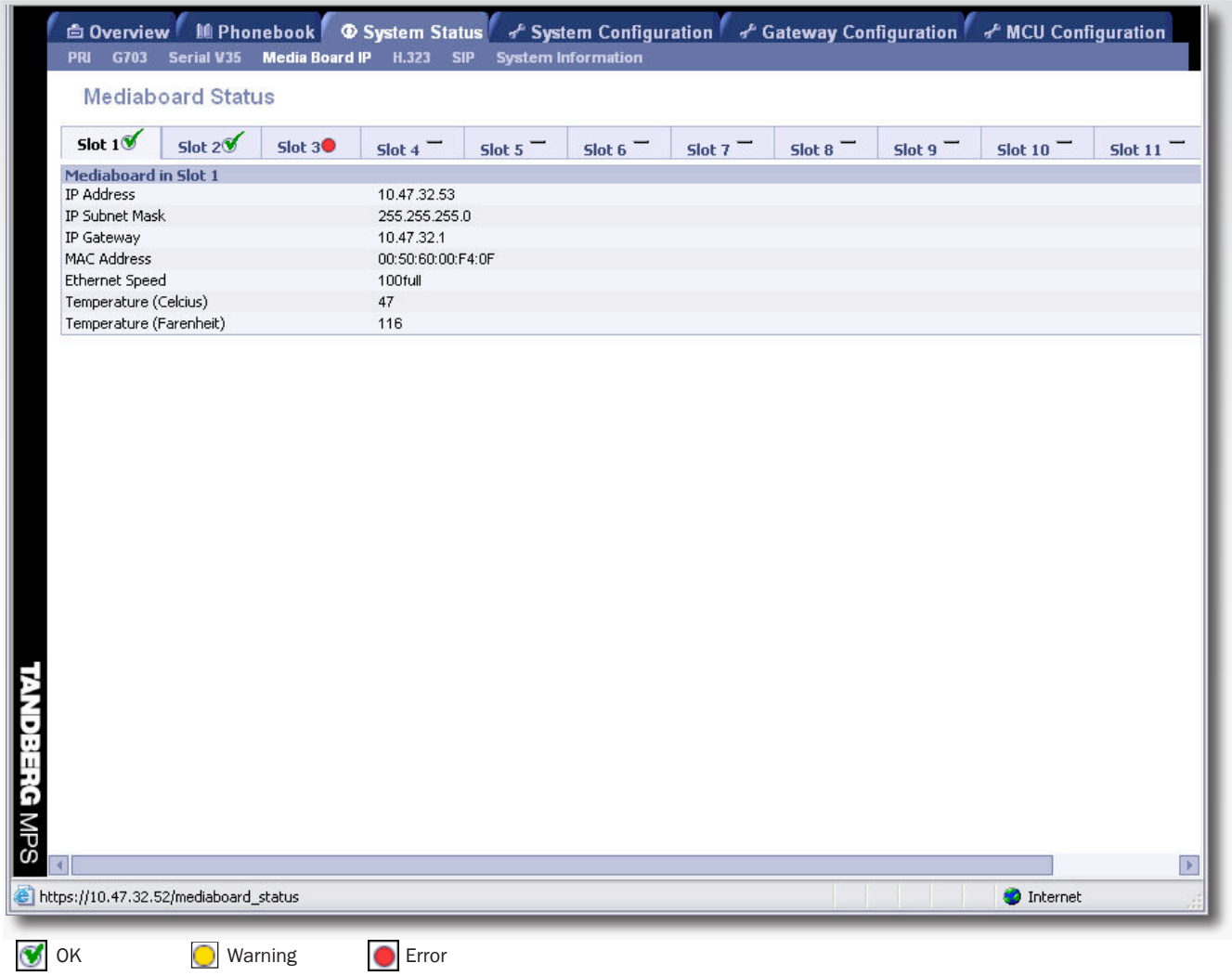
Temperature Celsius

Shows the temperature of the Media Processing Board in Celsius.

Temperature Fahrenheit

Shows the temperature of the Media Processing Board in Fahrenheit.

NOTE: Please ensure that there are no IP Address conflicts in your network. This is not detected by the MPS and may lead to loss of video and audio.



H.323 Status

IP Address

Shows the IP address of the TANDBERG MPS configured for Network #1.

H.323 Gatekeeper Status

Shows status and IP address of the Gatekeeper for Network#1, which the TANDBERG MPS is registered to.

- Inactive means the TANDBERG MPS is not registered to a Gatekeeper.
- Registering means the TANDBERG MPS is having problems registering with the selected Gatekeeper.
- Registered means the TANDBERG MPS is registered with the selected Gatekeeper.

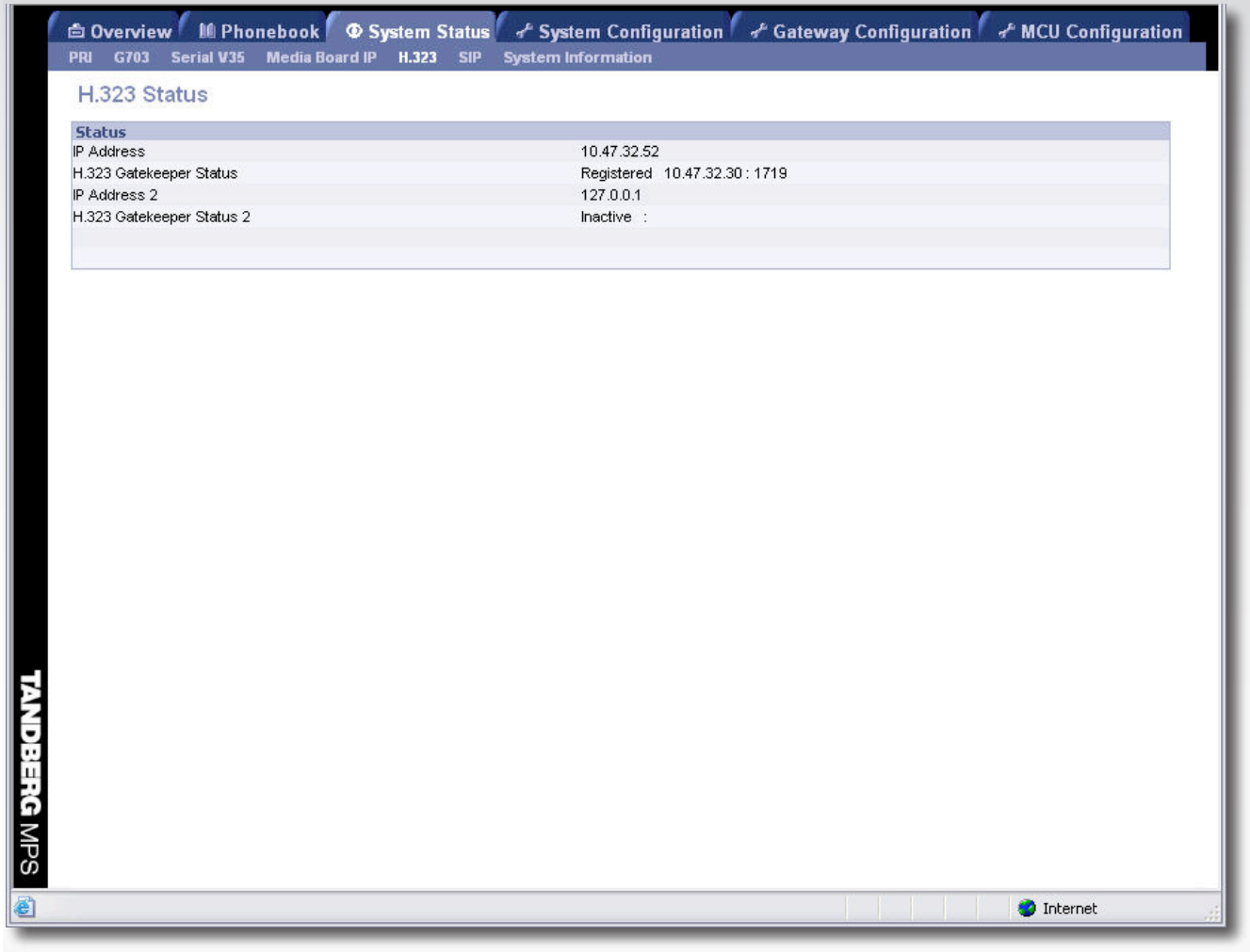
IP Address 2

Shows the IP address of the TANDBERG MPS configured for Network #2.

H.323 Gatekeeper Status 2

Shows status and IP address of the Gatekeeper for Network #2, which the TANDBERG MPS is registered to.

- Inactive means the TANDBERG MPS is not registered to a Gatekeeper.
- Registering means the TANDBERG MPS is having problems registering with the selected Gatekeeper.
- Registered means the TANDBERG MPS is registered with the selected Gatekeeper.



SIP Status

Mode - Shows if the SIP service is enabled or not.

Server Status

Show status towards the configured SIP server i.e. whether or not the MPS can find a server. Possible states are:

- INACTIVE - SIP server not in use
- DNS FAILED - DNS not set, external DNS server not found or SIP server address not found at DNS server
- TIMEOUT - SIP server not answering.
- UNABLE CONNECT TCP - Unable to connect to SIP server on TCP
- ACTIVE - Normal case, when server is found.
- UNKNOWN - Status unknown

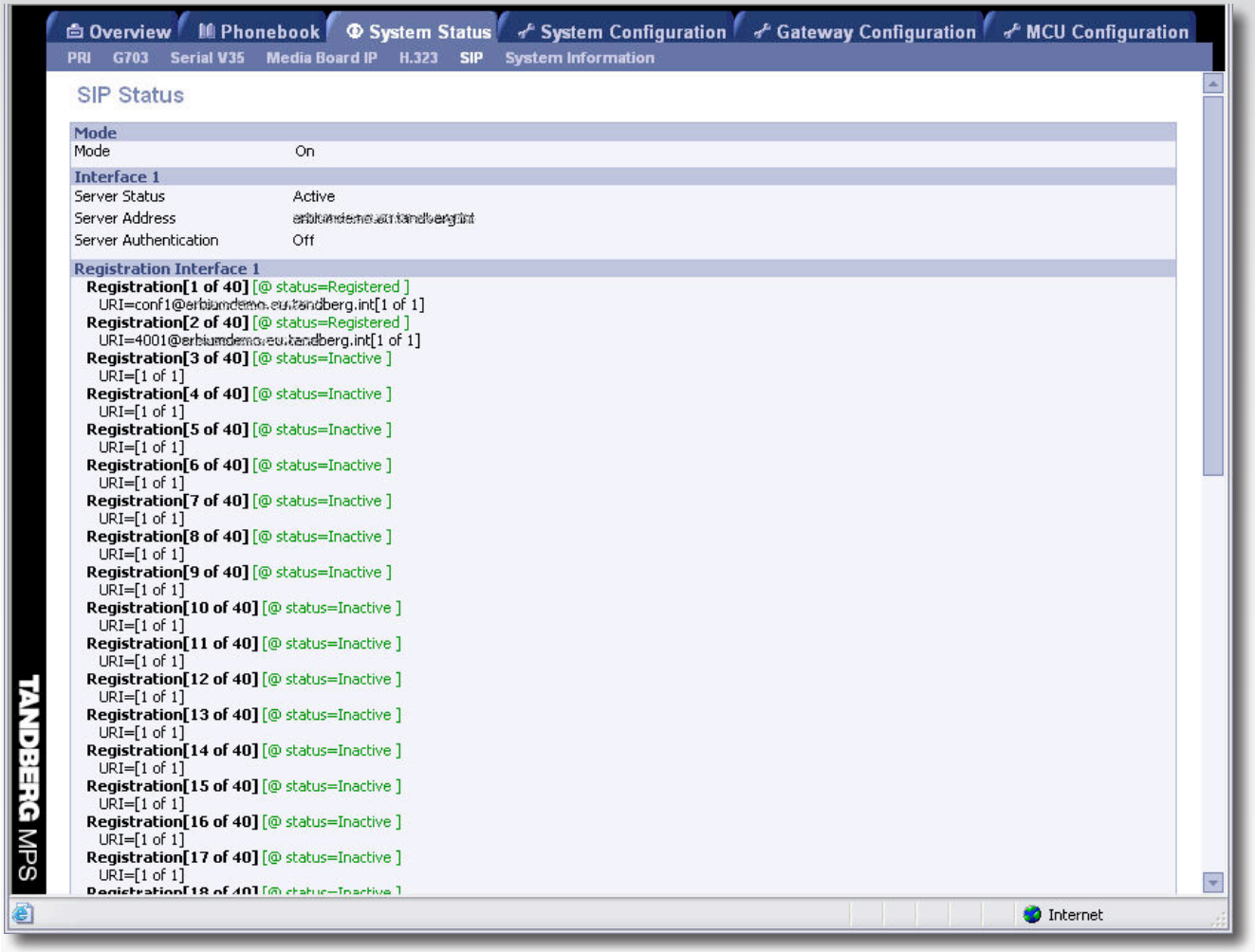
Server Address - Shows the connected SIP server address. If no server is selected, the normal state will be Off

Server Authentication - Shows if server NTLM or Digest Authentication is turned On or Off

Registration

One status line for each possible conference, all with their own unique SIP URI. Possible states are:

- INACTIVE - No SIP URI's have been logged on to server. In cases where the server Authentication is off, the different SIP URI's don't need to be registered, and will be reported as inactive
- REGISTERING - The server is registering the Conference URI
- REGISTERED - The Conference URI is registered on the SIP server.
- DEREGISTERING - The conference URI is deregistering from the SIP server.
- FAILED - The SIP server failed to register the URI



System Information

System Unit

- Product Type - Shows the TANDBERG product type
- Product ID - Shows the TANDBERG product id
- Up Time in Seconds - Shows the uptime since last boot in seconds
- Up Time - Shows the uptime since last boot in days, hours, minutes and seconds

Software

- Version - Shows the installed software version
- Name - Shows the installed software name
- Release Date - Shows the installed software release date

Configuration

- Telephony - Shows the total number of telephone options
- Video Telephony - Shows the total number of Video Telephone options
- Advanced Video Options - Shows the total number of Advanced Video options
- Encryption - Shows the total number of encryption sites
- PRI Ports - Shows the total number of PRI port options
- Serial Ports - Shows the total number of serial port options
- Maximum Bandwidth - Shows the maximum bandwidth installed

Managed Resources

- Conference - Shows the number of conferences available for scheduled conferences
- Fully Featured Conferences - Shows the number of fully featured conferences available for scheduled conferences
- Encoders - Shows the number of encoders available for scheduled conferences
- Telephony - Shows the number of telephone calls available for scheduled conferences
- Video Telephony - Shows the number of video calls available for scheduled conferences
- Advanced Video Option - Shows the number of advanced video calls available for scheduled conferences
- Encryption - Shows the number of encrypted calls available for scheduled conferences
- Max Bandwidth - Shows the maximum bandwidth available for scheduled conferences
- Max B Channel PRI - Shows the maximum number of B channels available for scheduled conferences

System Information, *cont...*

MCU Resources

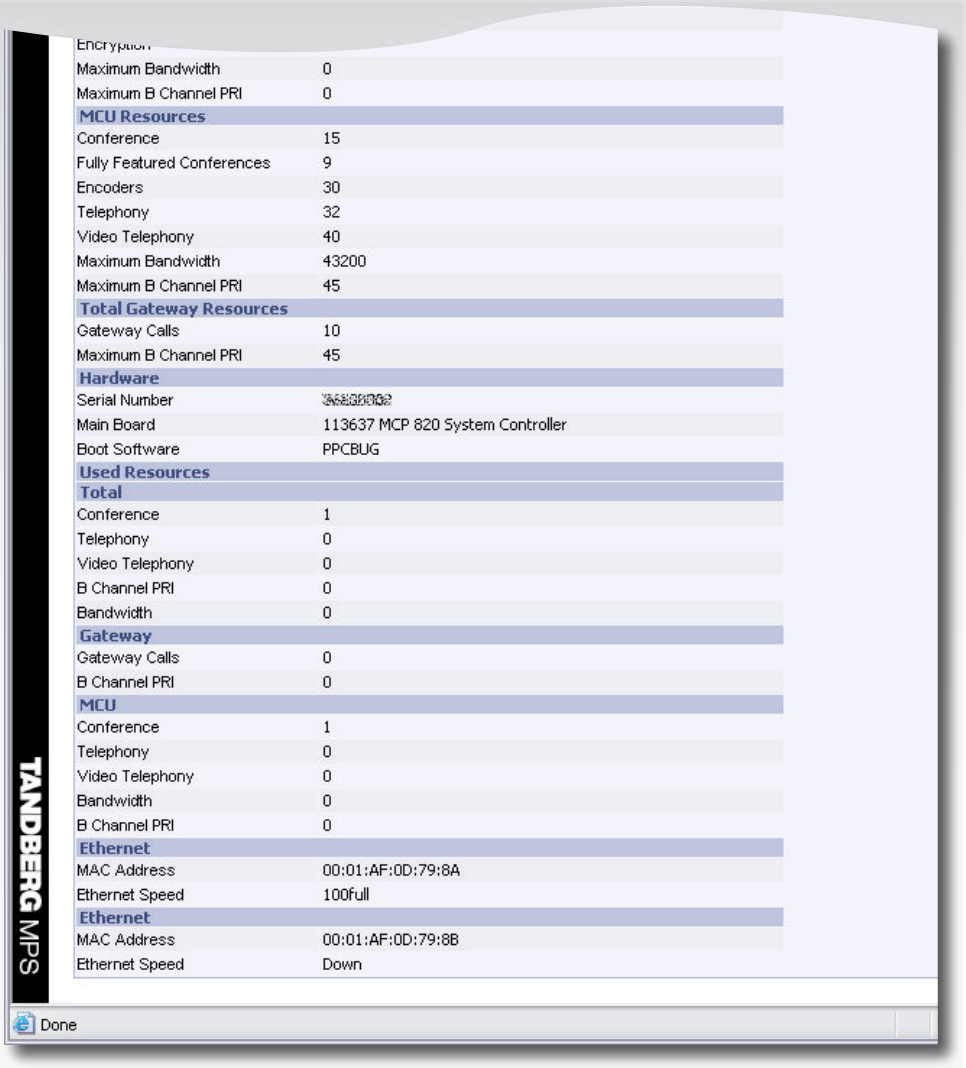
- Conference - Shows the total number of conferences available on the MPS
- Fully Featured Conferences - Shows the total number of fully featured conferences available on the MPS
- Encoders - Shows the total number of encoders available on the MPS
- Telephony - Shows the total number of telephone sites available on the MPS
- Video Telephony - Shows the total number of video telephone sites available on the MPS
- Maximum Bandwidth - Shows the maximum bandwidth available on the MPS
- Maximum B Channel PRI - Shows the maximum number of B channels available on the MPS

Total Gateway Resources

- Gateway Call - Shows the maximum number of Gateway calls available on the MPS
- Maximum B Channel PRI - Shows the maximum number of B channels available for the Gateway on the MPS

Hardware

- Serial Number - Shows the serial number for the MPS
- Main Board - Shows the mainboard id for the MPS
- Boot Software - Shows the boot software id for the MPS



System Information, *cont...*

Total

- Conference - Shows the total number of conference used on the MPS
- Telephony - Shows the total number of telephone calls used on the MPS
- Video Telephony - Shows the number of video calls used on the MPS
- B Channel PRI - Shows the total number of B channels used on the MPS
- Bandwidth - Shows the total bandwidth used on the MPS

Gateway

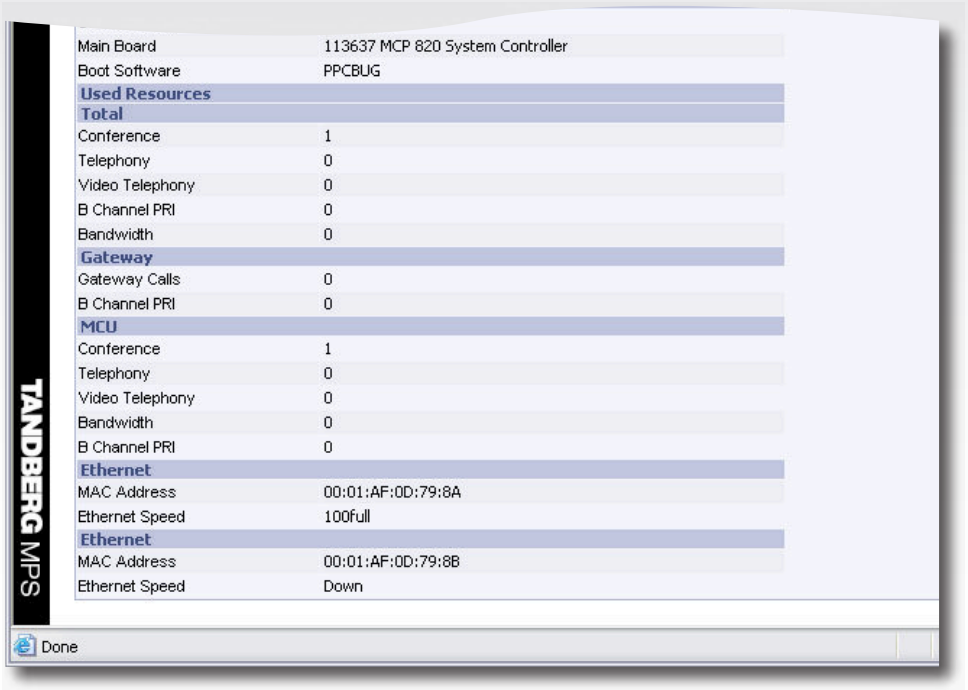
- Gateway Calls - Shows the number of Gateway calls used
- B Channel PRI - Shows the number of B channels used by the Gateway

MCU

- Conference - Shows the number of conferences used by the MCU
- Telephony - Shows the number for telephone calls used by the MCU
- Video Telephony - Shows the number of video calls used by the MCU
- Bandwidth - Shows the bandwidth used by the MCU
- B Channel PRI - Shows the number of B channels used by the MCU

Ethernet(s)

- MAC Address - Shows the Ethernet MacAddress(es)
- Ethernet Speed - Shows the Ethernet speed(s)



Chapter 5

System configuration

The System Configuration section contains information about how to configure and maintain the MPS.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

In this chapter...

H.320 Board Configuration

The Network Type must correspond with configurations set in the PRI Board Configuration if you have a PRI card or the G.703 Board Configuration if you have a leased line.

Slot (1 to n)

Select Slot 1 to 6 to configure the H.320 board(s). The illustration shows that Slot 1 and 3 are configured for H.320, while Slot 2, 4, 5 and 6 are not configured.

Network Type

Select which type of H.320 (ISDN) network each of the ISDN network cards are connected to:

PRI: ISDN Line

G.703: Leased line

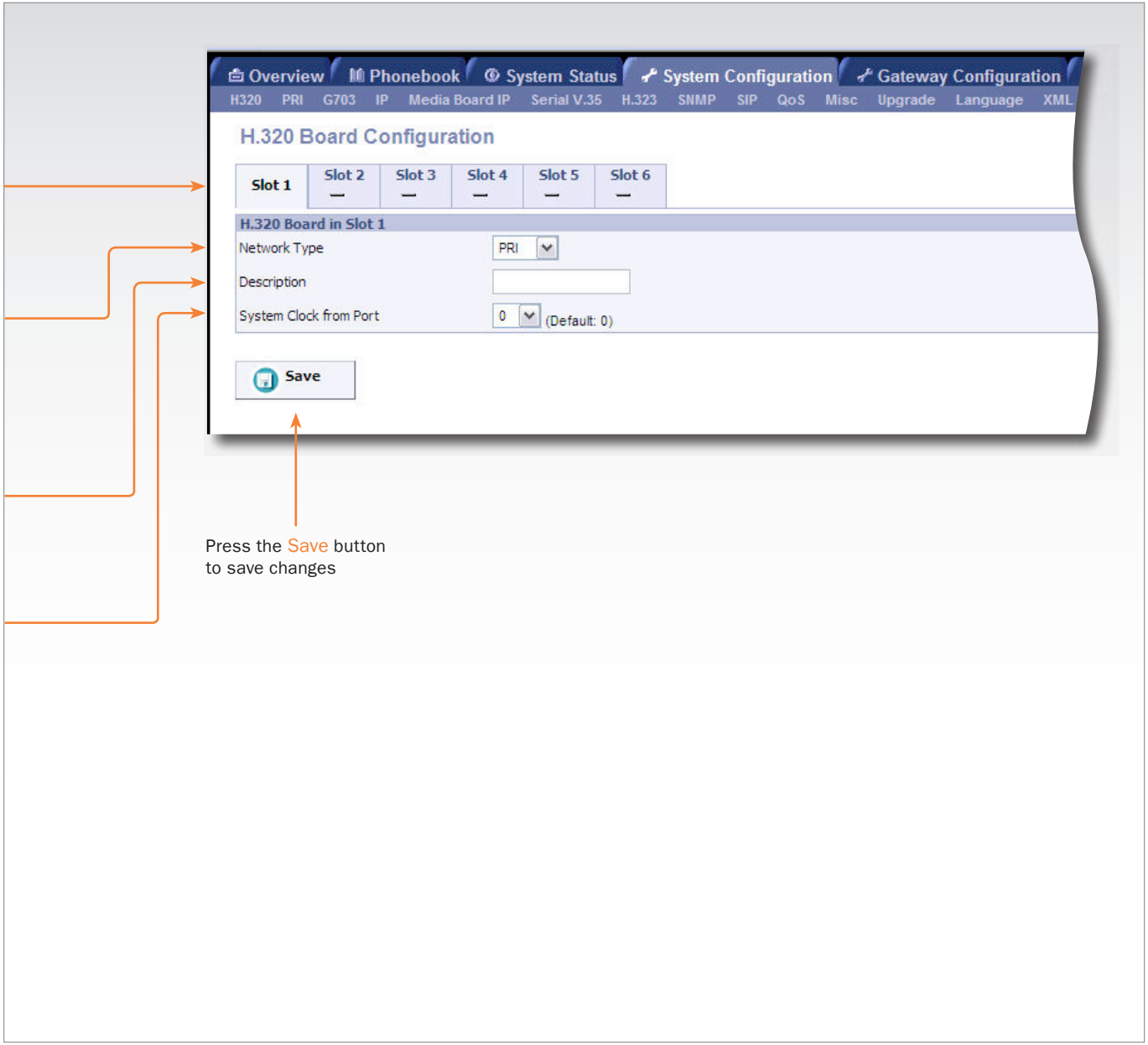
Description

Enter a compulsory textual description to easier identify each card.

System Clock from Port

Select which port to use as the System Clock source.

NB! If there are problems with this source, all H.320 calls may experience problems.



PRI Board Configuration

PRI Board Configuration, Slot 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

Send Complete

On: The TANDBERG MPS will send the ISDN message information element <Sending Complete> in the outgoing call Setup message. This is only required in a few countries.

Off: The TANDBERG MPS will not send <Sending Complete> (Default: Off).

Send Number

On: The TANDBERG MPS will send its own number to the far end. This is only useful when each conference have specified a Dial In number (Default: On).

Off: The TANDBERG MPS will not send its own number to the far end. Please note that the network may still send your number to the far end.

Parallel Dial

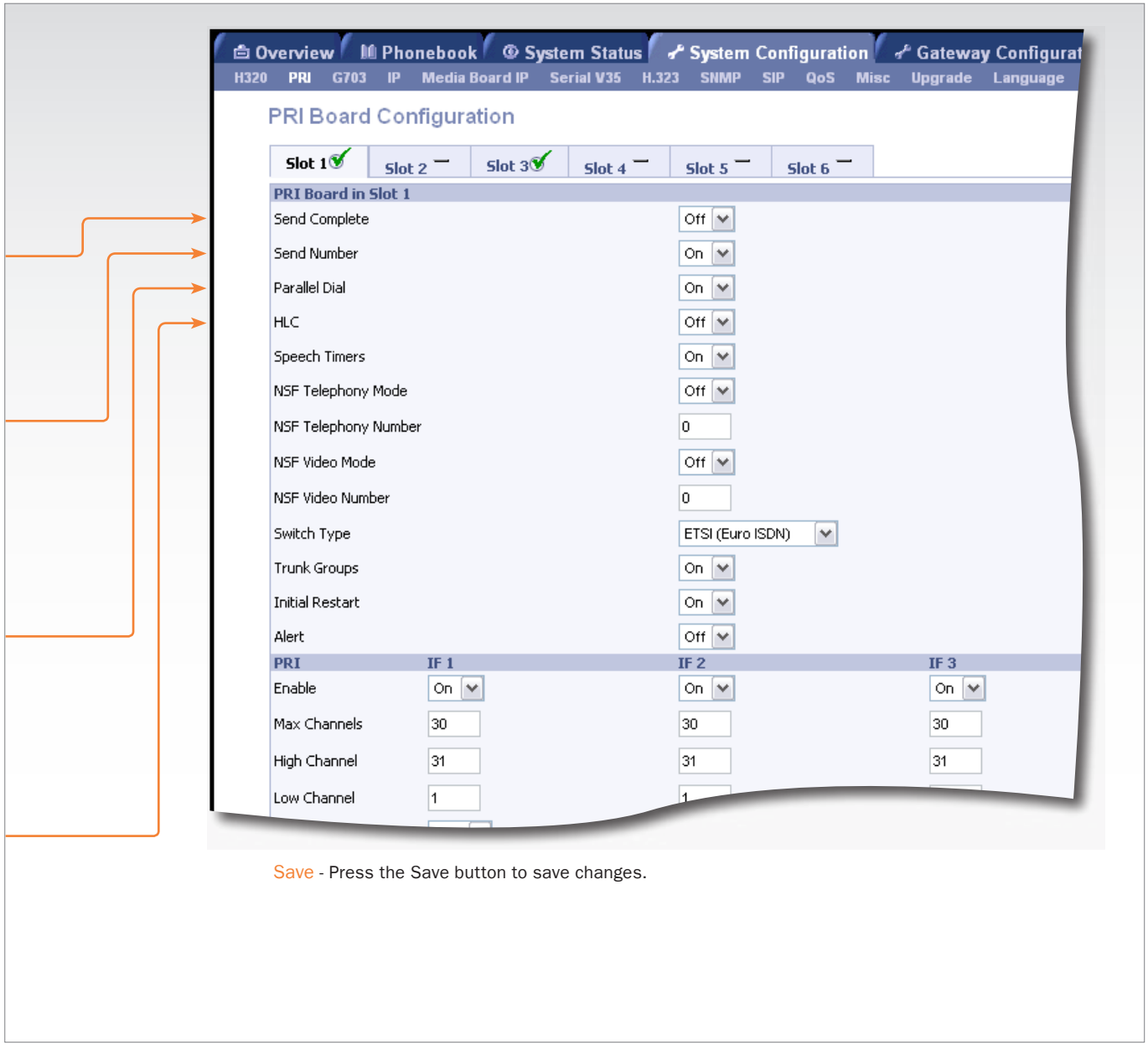
On: Channels will be dialed and connected in parallel when setting up a BONDING call (Default: On).

Off: Channels will be dialed sequentially, which may increase setup time.

HLC

On: Enables sending of HLC (High Level Compatibility) information element in setup message (video calls only).

Off: Disables sending of HLC information element in setup message (video calls only), (Default: Off).



Save - Press the Save button to save changes.

PRI Board Configuration, *cont...*

PRI Board Configuration, Slot 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

Speech Timers

On: If set to on, this will turn on Layer 3 timers T310, T304 and T301 for telephone calls (Default: On).

Off: If set to off, this will turn off Layer 3 timing.

NSF Telephony Mode

On: If set to on, NSF service code for telephone calls will be enabled.

Off: If set to off, NSF service code will be disabled (Default: Off).

NSF Telephony Number

Your network provider may require a service selection in your ISDN configuration.

Enter the Service code here. Valid NSF service codes are from 0 to 31 (National, AT&T or Japan/Taiwan ISDN).

NSF Video Mode

On: If set to on, NSF service code for video calls will be enabled.

Off: If set to off, NSF service code will be disabled (Default: Off).

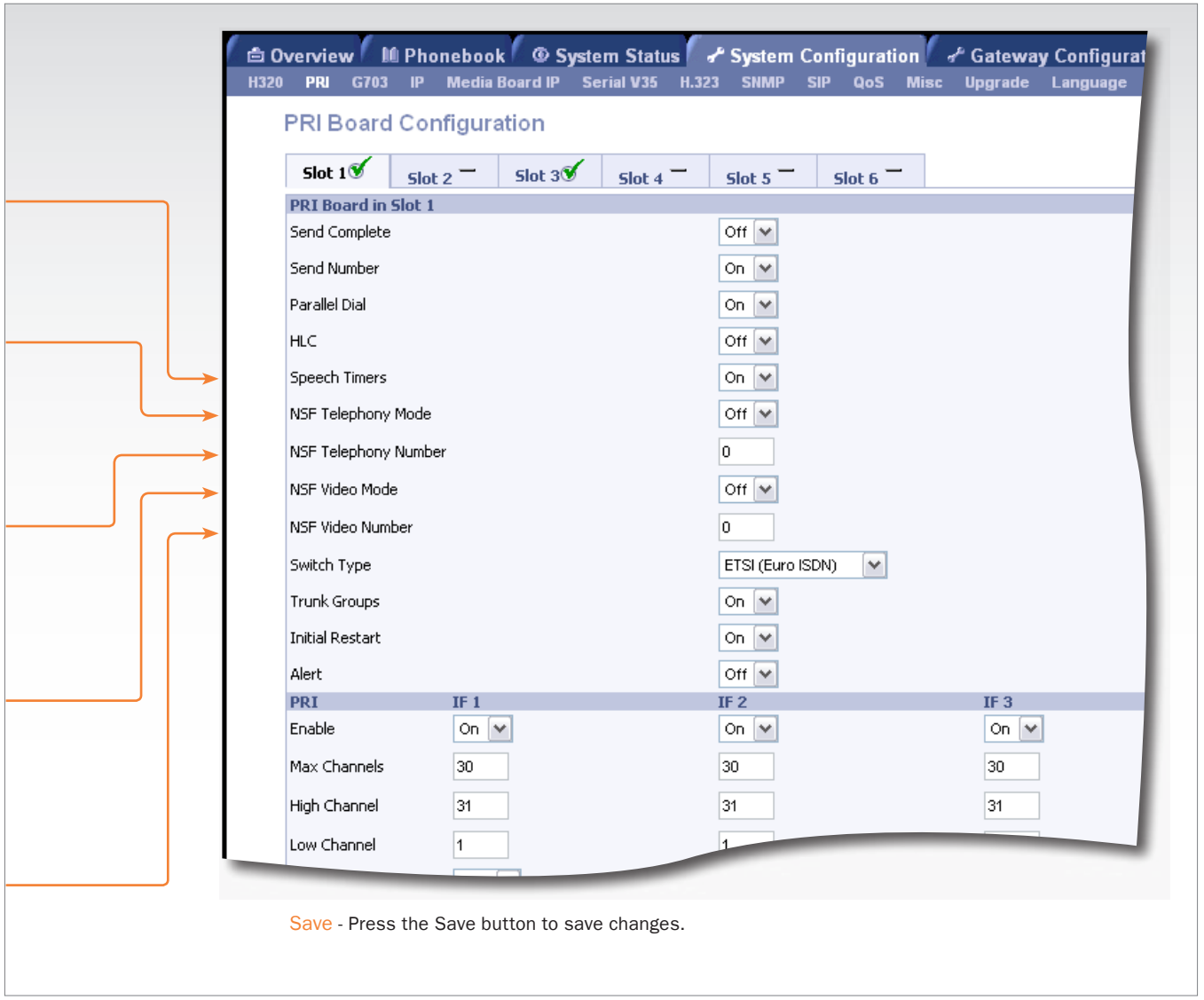
NSF Video Number

Your network provider may require a service selection in your ISDN configuration.

Enter the Service code here. Valid NSF service codes are from 0 to 31 (National, AT&T or Japan/Taiwan ISDN).

NSF - Network Specific Facilities

Read about NSF Service Codes in Appendices section.



Save - Press the Save button to save changes.

PRI Board Configuration, *cont...*

PRI Board Configuration, Slot 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

Switch Type - Select between the following PRI protocols:

- National ISDN
- AT&T Custom ISDN
- ETSI (Euro ISDN)
- Japan/Taiwan ISDN

NOTE! All PRI lines on the same E1/T1 Interface Card must use the same PRI Protocol.

Trunk Groups

On: When Trunk Groups is enabled, all of the PRI lines on the same E1/T1 Interface Card will use the same number range as specified for PRI 1, i.e. you will only need to specify the number range start and stop for port 1 and leave the fields for the other ports empty. (Default: On)

It is however necessary to set each individual port to On/Off in the Interface Configuration section below in this menu. See the next page.

Off: When set to Off, the Trunk Groups is disabled.

Initial Restart

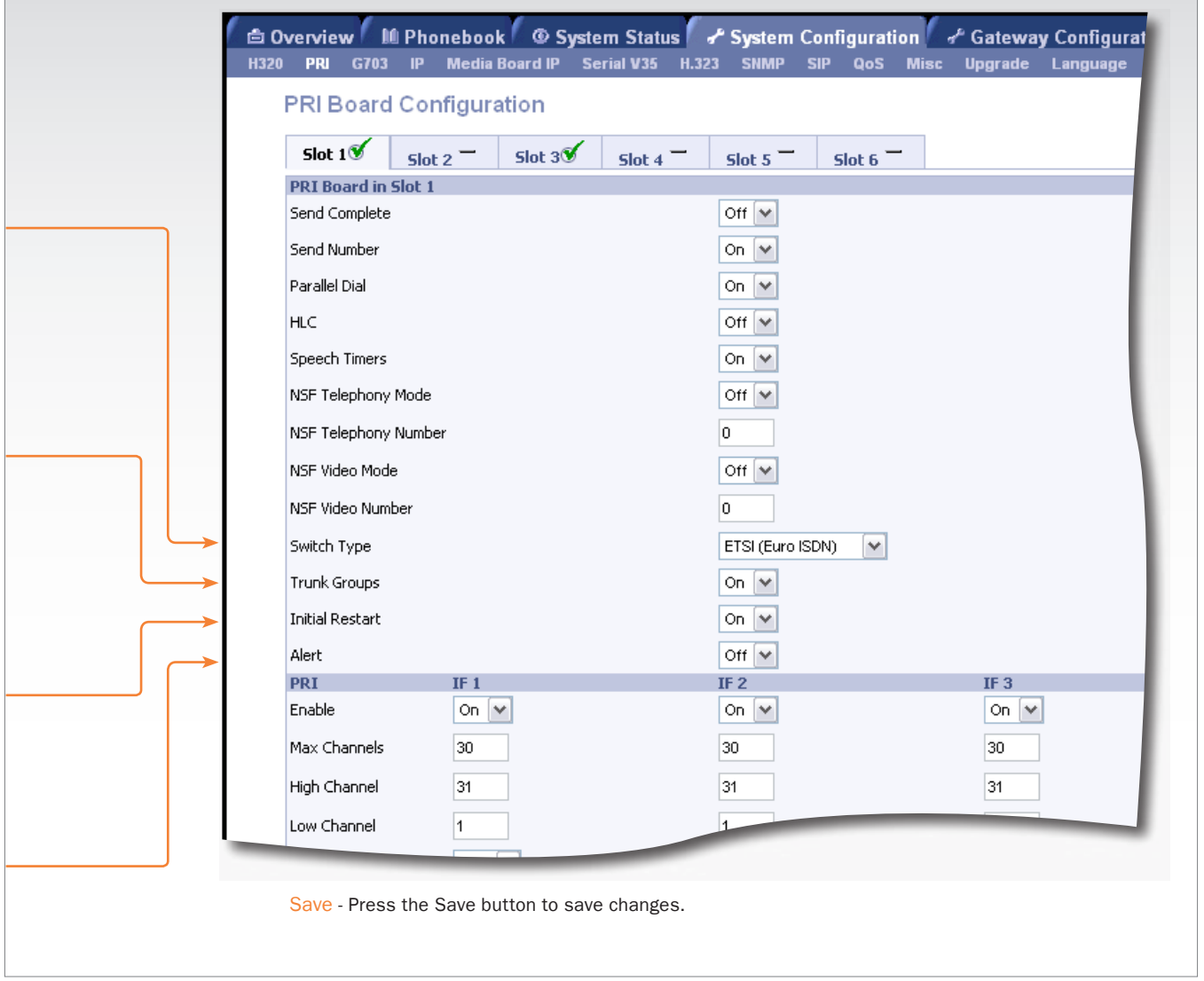
On: When set to On, the PRI interfaces will be reinitialized after boot (Default: On).

Off: When set to Off, the PRI interfaces will not be reinitialized after boot.

Alert

On: If set to On, the system will respond with an alert message to all incoming setup messages.

Off: If set to Off, the system will respond with an alert message only to the incoming setup message related to the initial channel (Default: Off).



Save - Press the Save button to save changes.

PRI Board Configuration, *cont...*

PRI Board Configuration, PRI Interface 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

PRI Interface Configuration

This section configures each of the PRI interfaces individually. There is one column for each PRI interface (IF 1, IF 2, IF 3, etc). However, if PRI Trunk Groups is enabled, the number range for PRI 1 will also apply for all the enabled PRI interfaces on the same E1/T1 Interface Card.

Enable Port

On: (default) If set to on, the PRI interface (IF #) is enabled. Off: If set to off, the PRI interface (IF #) is disabled.

Max Channels

Indicates the maximum number of B-channels the TANDBERG MPS is allowed to use for each of the PRI-interfaces.

- For E1 (ETSI/Euro ISDN), the maximum number of channels is 30. (Default: 30 for E1)
- For T1 (National ISDN and AT&T Custom), the maximum number of channels is 23. (Default: 23 for T1)

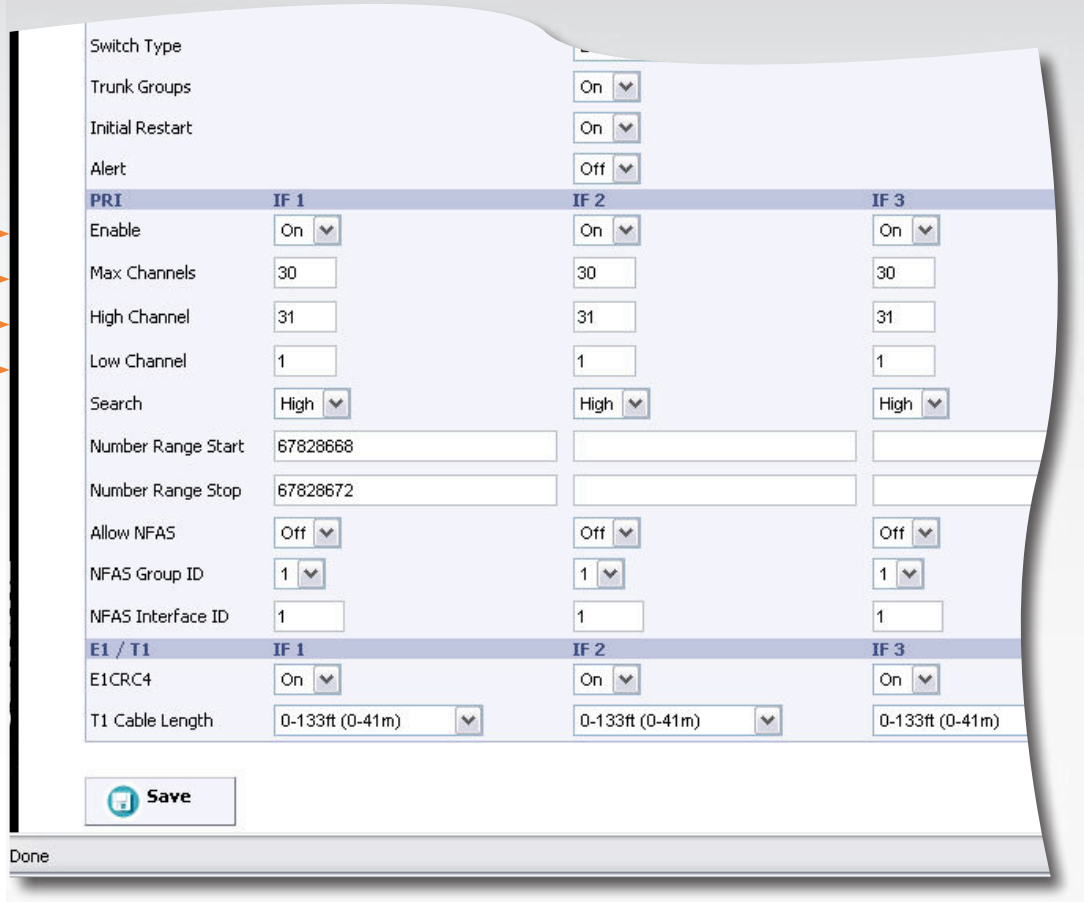
High Channel

Indicates the highest numbered E1/T1 B-channel the TANDBERG MPS is allowed to use for each PRI-line when selecting channels for outgoing calls. (Default: 23 for T1 and 31 for E1)

Low Channel

Indicates the lowest numbered E1/T1 B-channel the TANDBERG MPS is allowed to use for each PRI-line when selecting channels for outgoing calls. (Default: 1 for both T1 and E1).

- continued: PRI Board Configuration



Save - Press the Save button to save changes.

PRI Board Configuration, *cont...*

PRI Board Configuration, PRI Interface 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

Search

Specifies where the TANDBERG MPS will start searching for available B-channels for each PRI-line for an outgoing call.

High: If set to High, the TANDBERG MPS will start to search for available B-channels at the highest numbered B-channel (Default: High)

Low: If set to Low, the TANDBERG MPS will start searching for available B-channels at the lowest numbered B-channel.

The MPS tries to keep the H0-channels free as long as possible and will therefore not start searching at the lowest numbered channel when set to Low.

Number Range Start

The PRI lines connected to the TANDBERG MPS should have at least one number each, to allow dial in from ISDN. If the PRI line has a range of numbers, the start number must be entered here. (Default: Empty)

The number range must be inclusive. All numbers in the range may be used by the TANDBERG MPS for callback numbers, so the local ISDN switch must route all of these numbers to the configured PRI. Please contact your IT manager or ISDN service provider to obtain these numbers.

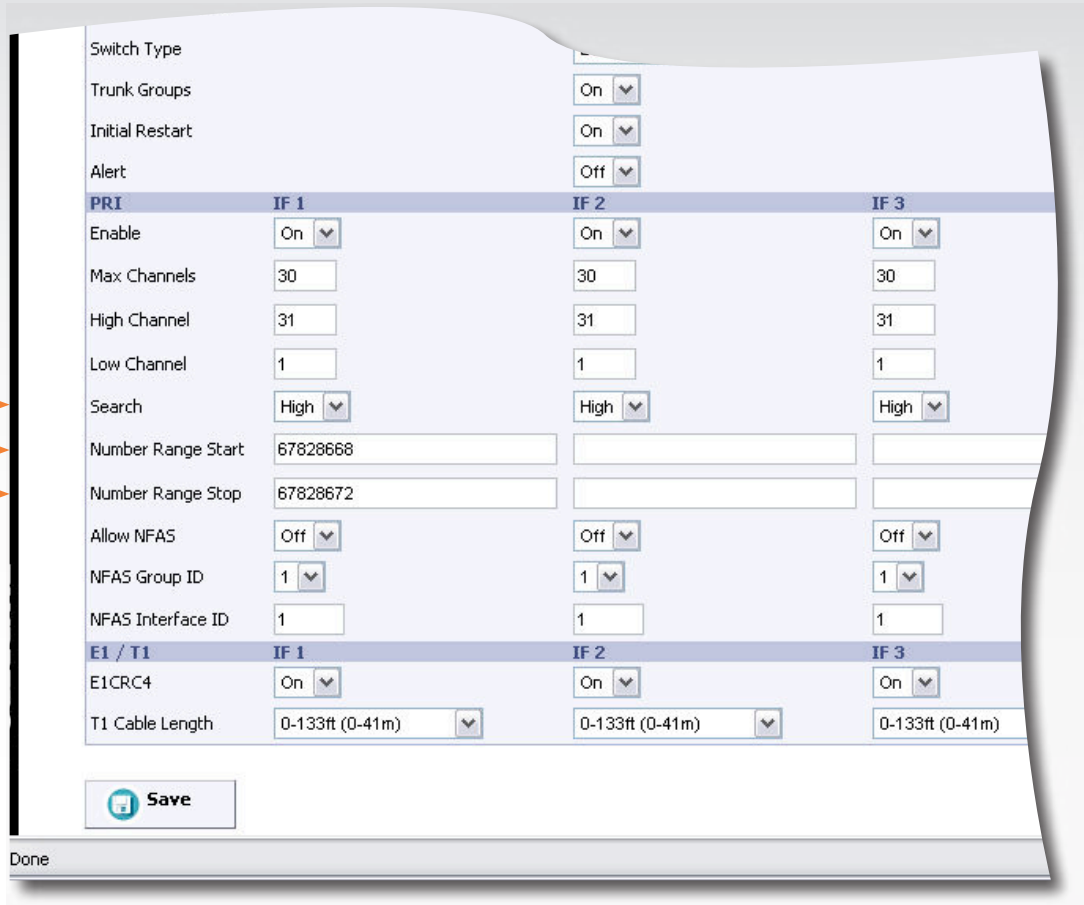
It is only necessary to enter the digits indicating the range. If the range is 67828669 to 67828699, then just enter 8669. Maximum amount of digits is 24.

Number Range Stop

Here is where the last number in the PRI number range is entered. If the range is 67828669 to 67828699, then just enter 8699. (Default: Empty)

NOTE: When receiving a call with the Gateway configured with only one number in the PRI number range, all other incoming calls will be blocked until the ongoing call is established. This is because the network will have no additional number available for the second call while setting up the call. When the call is established the number will be available and any incoming call will connect as normal.

- continued: PRI Board Configuration



Save - Press the Save button to save changes.

PRI Board Configuration, *cont...*

PRI Board Configuration, PRI Interface 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

Allow NFAS

On: If set to On, the Allow NFAS is enabled.

Off: If set to Off, the Allow NFAS is disabled.

Non-Facilities Associated Signalling (NFAS), is an ISDN feature for sharing one ISDN D-channel across multiple ISDN PRI interfaces.

- The NFAS is only available on T1 networks
- Backup D channels are not currently supported.

Example: If we use NFAS to share one D channel across eight PRI interfaces, we will gain seven extra B channels over a configuration that used one D channel per PRI interface. Additionally we may save expenses related to seven D channels.

INFO: The Initial Restart setting should be set to Off when the Allow NFAS setting is enabled.

NFAS Group ID

It is possible to have 4 NFAS groups per card, where each group may cover two or more PRI interfaces. Enter a value between 1 and 4.

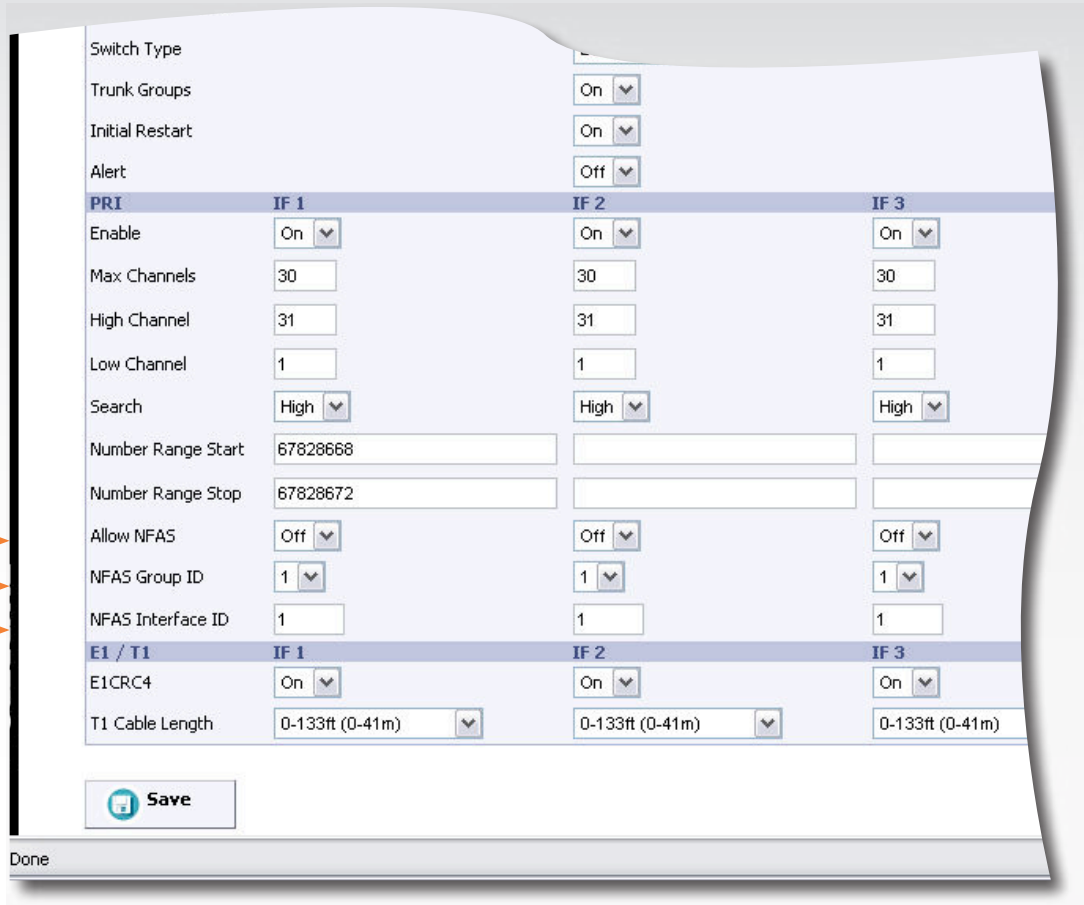
INFO: This setting will only take effect when Allow NFAS is set to On.

NFAS Interface ID

Each interface in an NFAS group must be assigned an NFAS Interface ID.

- These IDs must be the same as the ones used on the network side of the interface.
- Enter a value between 0 and 127.
- The interface carrying the D channel for signalling must be set to 0.

- continued: PRI Board Configuration



Save - Press the Save button to save changes.

PRI Board Configuration, *cont...*

PRI Board Configuration, E1/T1 Interface 1–n

The Network Type must correspond with configurations set in the PRI Board Configuration.

This section describes how to configure each of the E1/T1 interfaces, with one column for each interface (IF1, IF2,IF3, etc).

However, if PRI Trunk Groups is enabled, the number range for PRI 1 will also apply for all the enabled PRI interfaces on the same E1/T1 Interface Card.

E1 CRC4

Used for most E1-PRI configurations.

On: If set to On, the E1 CRC4 is enabled.

Off: If set to Off, the E1 CRC4 is disabled. If your network equipment does not support this feature, turn it Off.

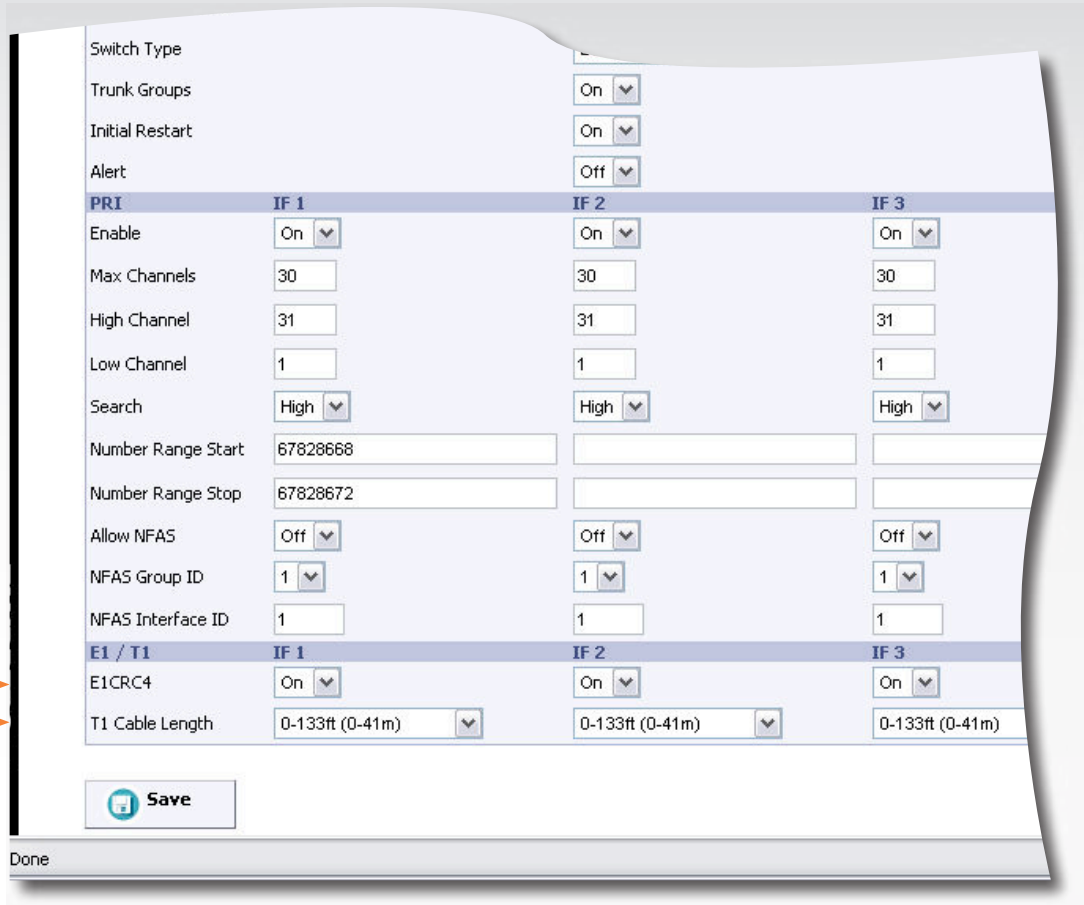
T1 Cable Length

Specifies the cable length between the TANDBERG MPS and the CSU for each of the PRI lines (only valid for T1 networks).

Possible values are:

- Range1: 0 to 133 feet
- Range2: 133 to 266 feet
- Range3: 266 to 399 feet
- Range4: 399 to 533 feet
- Range5: 533 to 655 feet

- continued: PRI Board Configuration



Save - Press the Save button to save changes.

G.703 Board Configuration

PRI Board Configuration

Ensure that Network Type is set to G.703 for the connected ISDN cards in the PRI Network Configuration.

G.703 Board

Leased Line is a non-dialling protocol. Should be used when:

1. One codec is connected in a point to point connection
2. Through an ISDN switch supporting G.703 Leased Line mapping.

NOTE! It may be necessary to specify the call rate explicitly when dialling a G.703 call id to ensure that the correct call rate is used.

Slot (1 to n)

Select Slot 1 to 6 to configure the G.703 (Leased Line) board. The picture shows one G.703 board installed in Slot 3.

OK Warning Error

Physical Layer

Select E1 (max 30+1 channels) or T1 (max 23+1 channels) to match your ISDN network.

Interface (1 to 8)

The interface number refers to the port number/G.703 interface of the Network Interface Card.

Enable

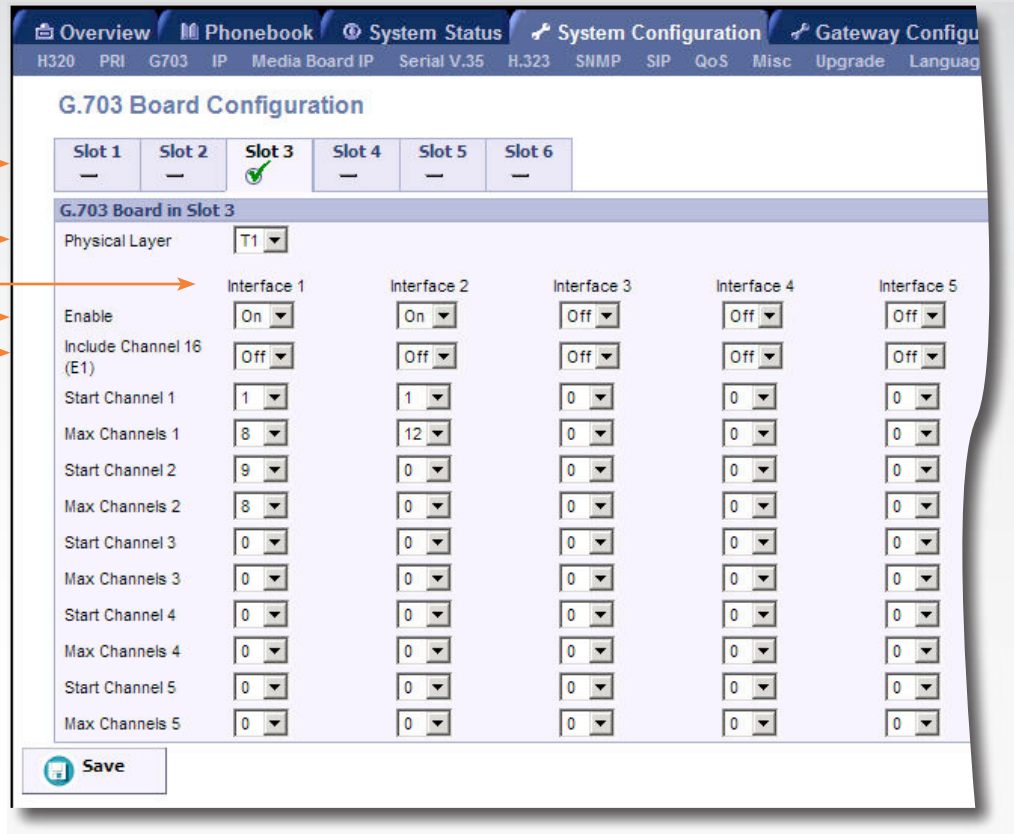
On: If set to On, the port is enabled for G.703 use.

Off: If set to Off, the port is disabled for G.703 use.

Include Channel 16 (E1)

On: Includes channel 16 when configured as E1.

Off: Disables channel 16 when configured as E1 .



Save - Press the Save button to save changes.

G.703 Board Configuration, *cont...*

PRI Board Configuration

Ensure that Network Type is set to G.703 for the connected ISDN cards in the PRI Network Configuration.

G.703 Board

Leased Line is a non-dialling protocol. Should be used when:

1. One codec is connected in a point to point connection
2. Through an ISDN switch supporting G.703 Leased Line mapping.

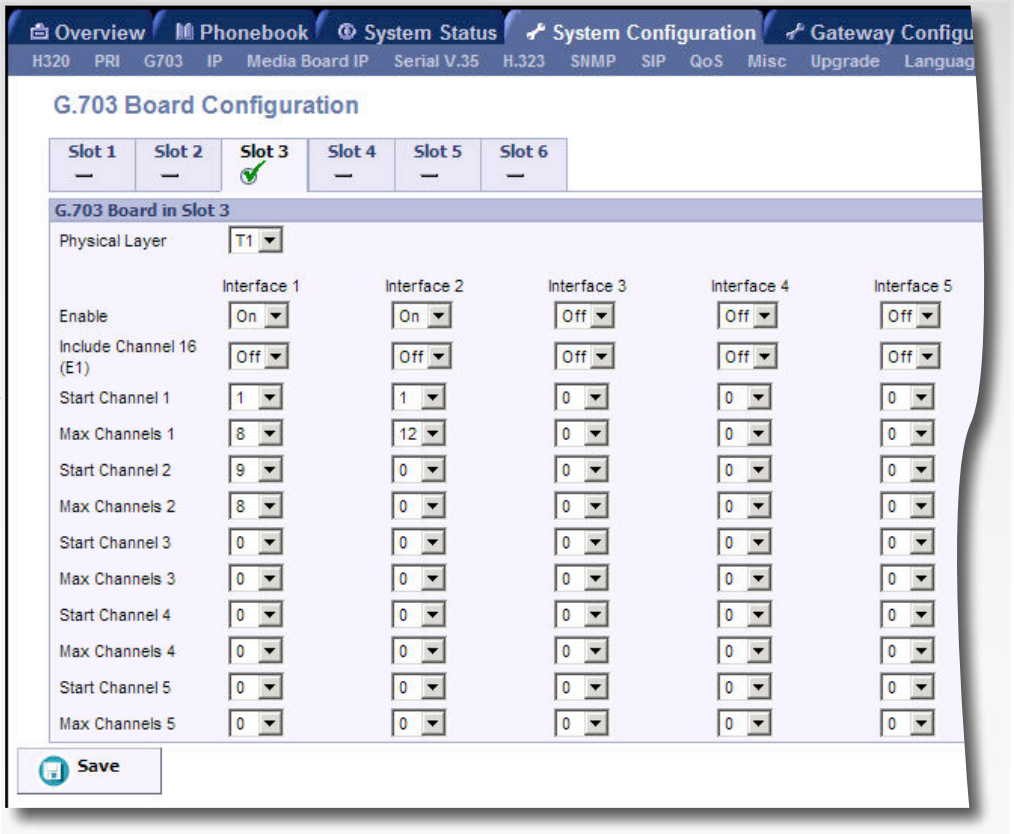
Channel (1 to 5)

For each G.703 Leased Line interface it is possible to define one or more calls. Each call is identified by a number ("call-id"), a start channel and max number of channels. The maximum number of possible calls is 5.

Start Channel (1 to 5)	Max Channels (1 to 5)
E1: Start channel <1..31>	Max channels <0 (Off), 1..31>
T1: Start channel <1..24>	Max channels <0 (Off), 1..24>

Table: Example with four different call settings:

	5 Calls	4 Calls	3 Calls	2 Calls
Start Channel 1	1	1	1	1
Max Channels 1	4 (256 k)	6 (384 k)	8 (512 k)	12 (768 k)
Start Channel 2	5	7	9	13
Max Channels 2	4 (256 k)	6 (384 k)	8 (512 k)	12 (768 k)
Start Channel 3	9	13	17	
Max Channels 3	4 (256 k)	6 (384 k)	8 (512 k)	
Start Channel 4	13	19		
Max Channels 4	4 (256 k)	6 (384 k)		
Start Channel 5	17			
Max Channels 5	4 (256 k)			



Save - Press the Save button to save changes.

IP Configuration

IP Configuration Interface 1 and 2

As a minimum you must configure Interface 1. If your TANDBERG MPS is connected to two different IP-networks, you must configure both Interface 1 and Interface 2.

INFO: Before you can access the MPS from a web browser the initial configuration, of a static IP address for the System Controller Board, is done via the LCD.

Address

The IP Address defines the network address of the System Controller Board. Your LAN administrator will provide you with the correct address for this field.

Subnet Mask

The IP Subnet Mask defines the type of network. Your LAN administrator will provide the correct value for this field.

Gateway

The IP Gateway defines the Gateway address. Your LAN administrator will provide the correct value for this field.

Ethernet Speed

Auto: The MCU will automatically detect the speed/duplex on the LAN.

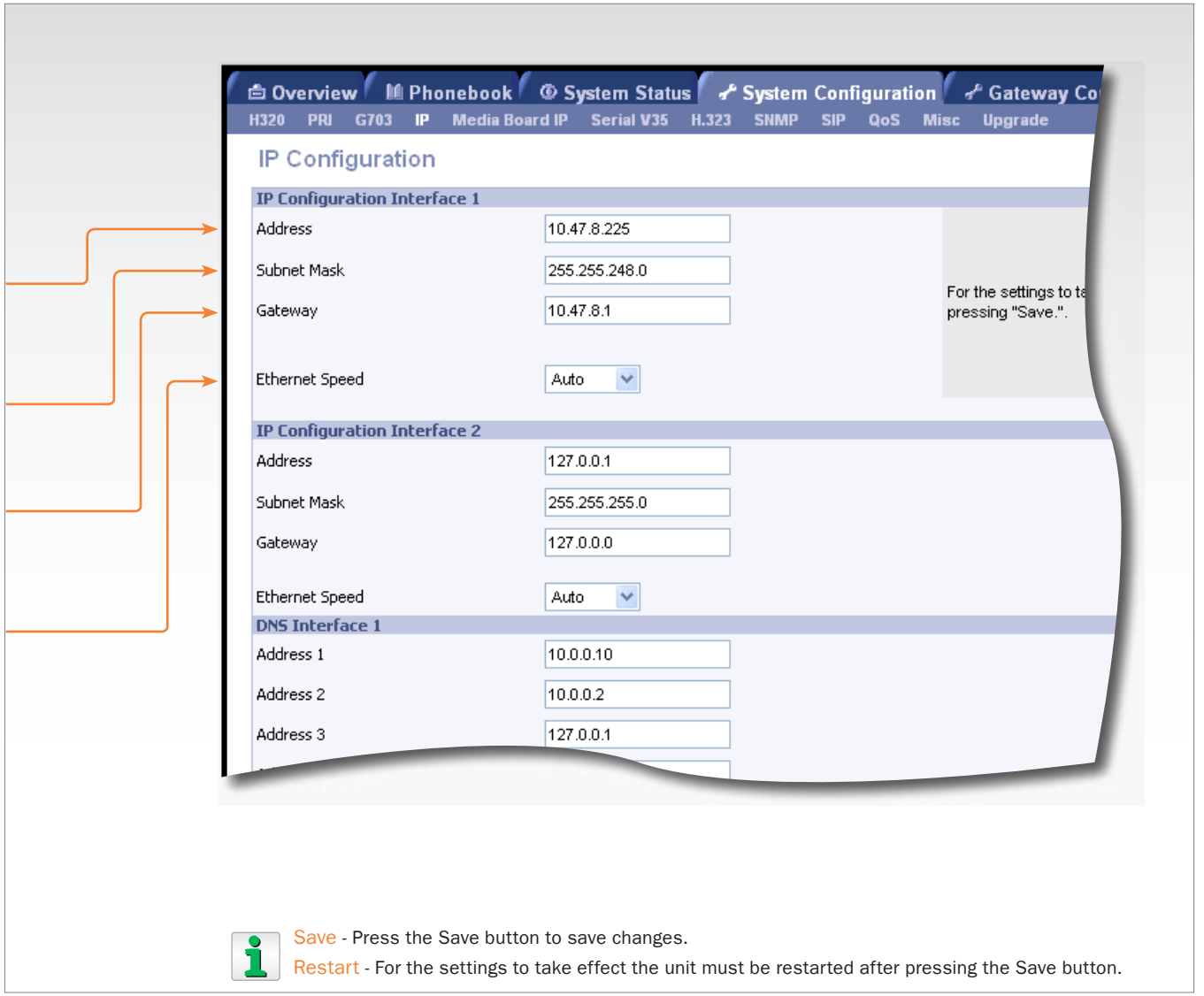
10Half: The MCU will connect to the LAN using 10 Mbps/Half Duplex.

10Full: The MCU will connect to the LAN using 10 Mbps/Full Duplex.

100Half: The MCU will connect to the LAN using 100 Mbps/Half Duplex.

100Full: The MCU will connect to the LAN using 100 Mbps/Full Duplex.

If forcing the TANDBERG MPS to anything other than Auto then the Ethernet switch must also be forced to match.



Save - Press the Save button to save changes.

Restart - For the settings to take effect the unit must be restarted after pressing the Save button.

IP Configuration, *cont...*

DNS Interface 1 and 2

DNS Server defines the network addresses for DNS servers, interface 1 and 2. Up to 5 addresses may be specified.

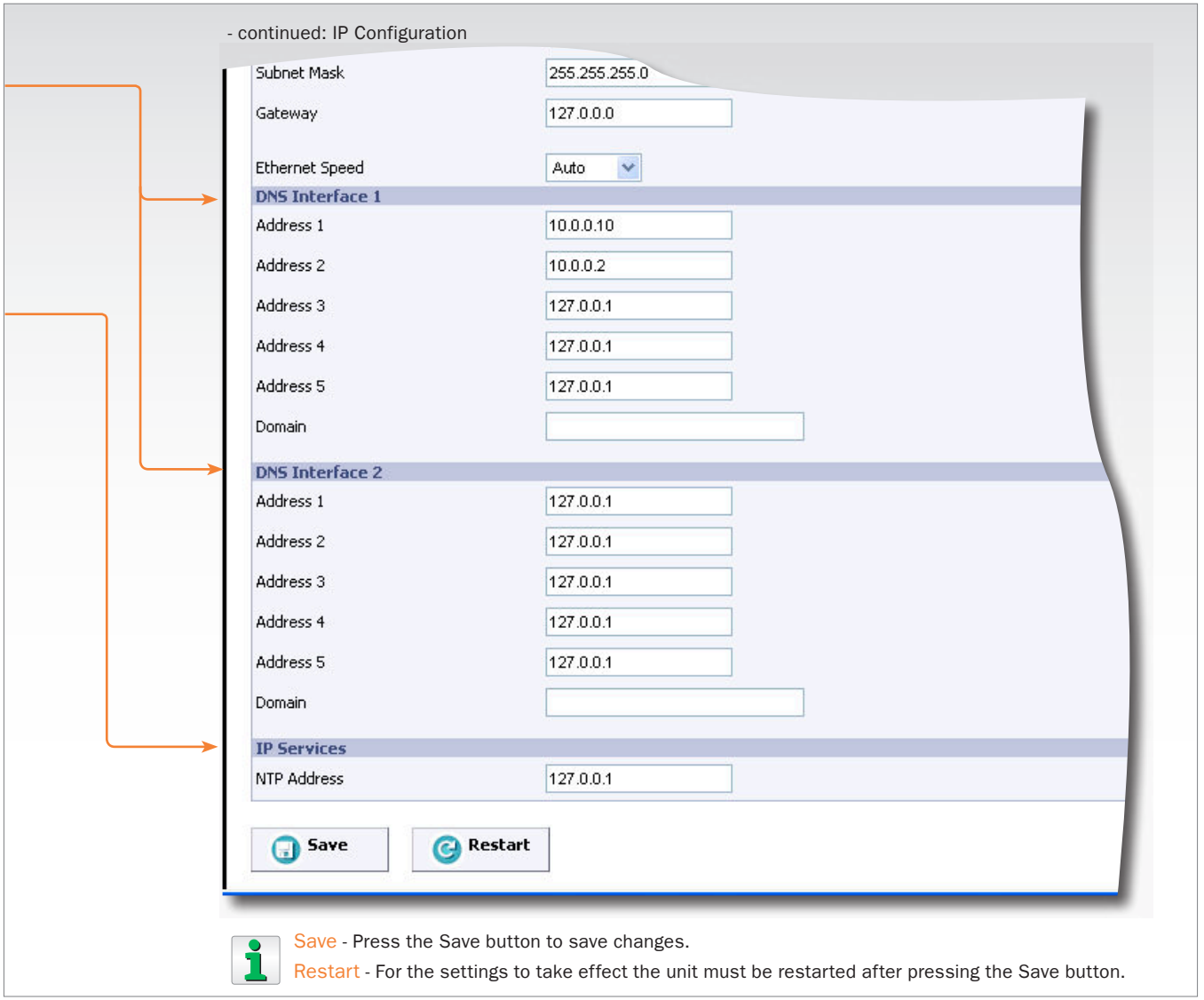
If the network addresses are unknown, please contact your LAN administrator or Internet Service Provider.

The DNS Domain Name is the default domain name suffix which is added to unqualified names.

IP Services

This is the NTP time server address from which the system can get updated time information.

- The Network Time Protocol (NTP) is used to synchronize the time of the system to a reference time server.
- The NTP IP setting holds the IP address to a time source where the system can get the exact time.
- The time server will subsequently be queried every 24th hour for time updates.



Save - Press the Save button to save changes.

Restart - For the settings to take effect the unit must be restarted after pressing the Save button.

Media Board Configuration

Media Board Configuration - Slot (1 to n)

Select Slot 1 to 12 to configure the Media Board(s). The picture shows that Media Boards are installed Slot 1, 2 and 3.

OK Warning Error

Network ID

1: The Media Processing Board is connected to the same IP-network as the System Controller Board Interface 1.

2: The Media Processing Board is connected to the same IP-network as the System Controller Board Interface 2.

Ethernet Speed

Auto: The MCU will automatically detect the speed/duplex on the LAN.

10Half: The MCU will connect to the LAN using 10 Mbps/Half Duplex.

10Full: The MCU will connect to the LAN using 10 Mbps/Full Duplex.

100Half: The MCU will connect to the LAN using 100 Mbps/Half Duplex.

100Full: The MCU will connect to the LAN using 100 Mbps/Full Duplex.

If forcing the TANDBERG MPS to anything other than Auto then the Ethernet switch must also be forced to match.

Address

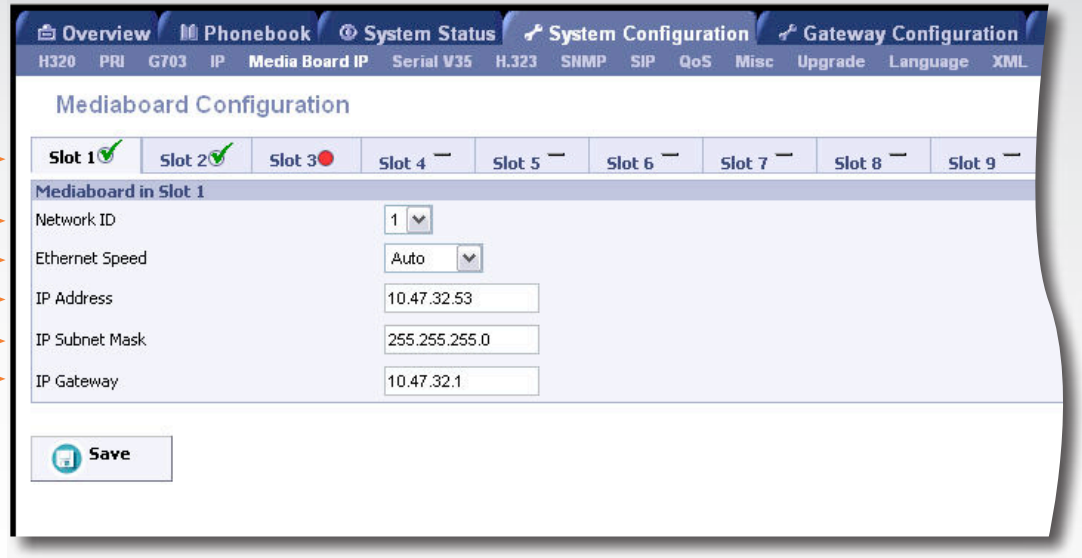
The IP Address defines the network address of the Media Board. Your LAN administrator will provide you with the correct address for this field.

Subnet Mask

The IP Subnet Mask defines the type of network. Your LAN administrator will provide the correct value for this field.

Gateway

The IP Gateway defines the Gateway address. Your LAN administrator will provide the correct value for this field.



Save - Press the Save button to save changes.

NOTE: Please ensure that there are no IP Address conflicts in your network. This is not detected by the MPS and may lead to loss of video and audio.

Serial V.35 Board Configuration

Slot (1 to n)

Select Slot 1 to 6 to configure the Serial V.35 Board(s). The picture shows that Serial V.35 Boards installed on Slot 2.

OK Warning Error

Internal Clock Rate

Sets the Internal Clock Rate which the MPS will provide to all ports when Clocking is set to Internal. Can be set to a clock rate from 64kbps up to 1920kbps.

Restrict

This Restrict setting will impact the Internal Clock Rate only. In addition, each port using internal clocking must also have the Restrict set to on.

On: Setting Restrict to On will increment the Internal Clock Rates in steps of 56kbps.

Off: Setting Restrict to Off will increment the Internal Clock Rates in steps of 64kbps.

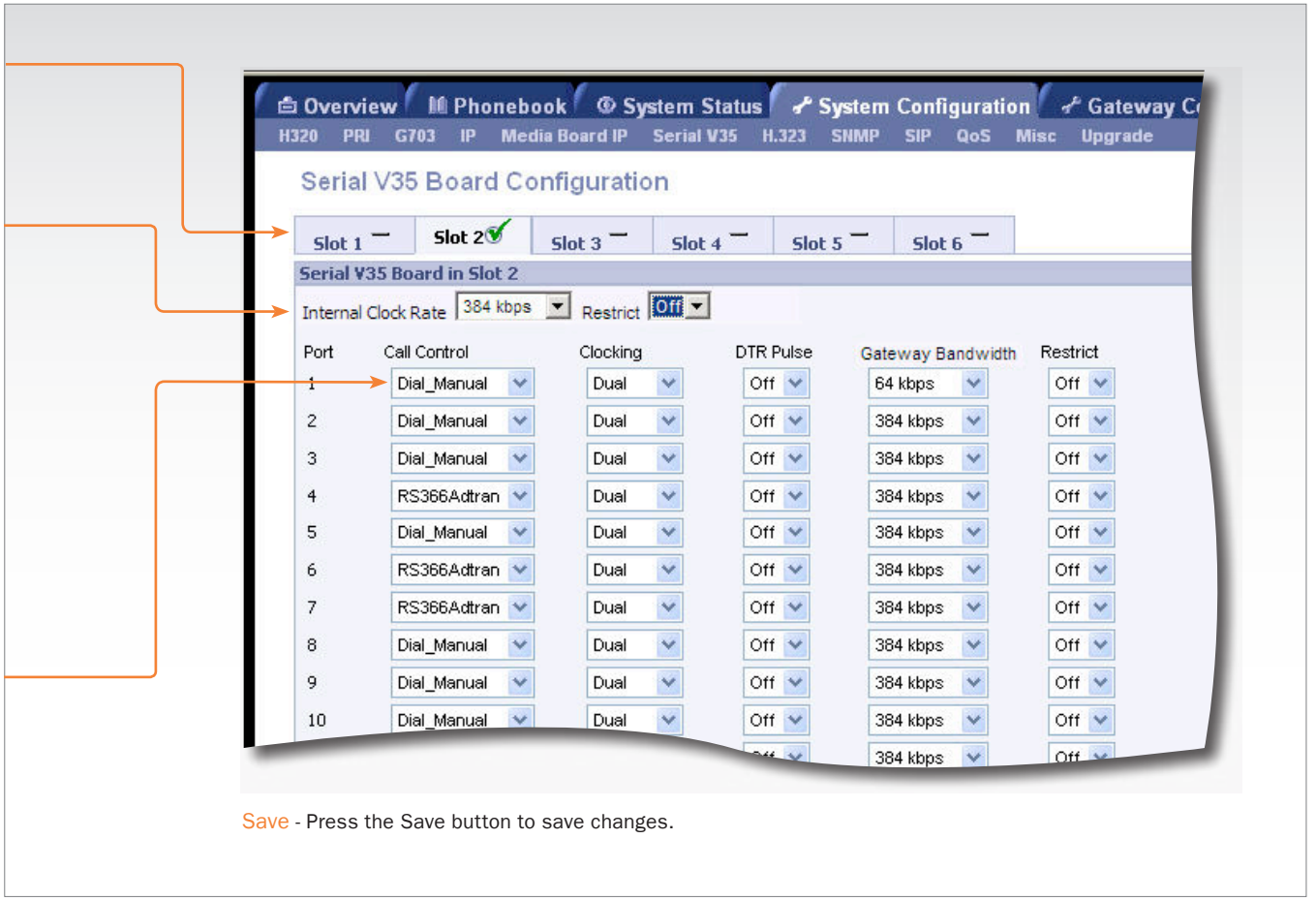
Call Control, port 1-32

Sets the call control settings to each port

Dial Manual: Select Dial Manual when no handshake signals are available or when using RS366 with other IMUX and the external equipment requires a constantly connected line. (Default: Dial Manual)

RS366 ADTRAN: Select RS366 ADTRAN only when connected to an ADTRAN ISU 512.

- The ADTRAN ISU 512 offers extra usability when dialling RS366 via an ADTRAN ISU 512 IMUX.
- This dialling scheme will map the call type and bandwidth selection to ADTRAN ISU 512 specific suffixes to the dialled number.



Save - Press the Save button to save changes.

Serial V.35 Board Configuration, *cont...*

Slot (1 to n)

Select Slot 1 to 6 to configure the Serial V.35 Board(s). The picture shows that Serial V.35 Boards installed on Slot 2.

OK Warning Error

Clocking, port 1-32

Sets the clocking mode to each port

Dual: When using RS449, RS530, and V.35 the external equipment provides two clock signals, one for transmit and one for receive. The difference between RS449, RS530 and V.35 is the cable only (Default: Dual).

Single: Used when the external equipment provides one common clock signal, X.21, for both transmit and receive.

Internal: Used when the external device has no clocking. Clocking will be provided by the MPS according to the Internal Clock Rate settings.

DTR Puls, port 1-32

On: Set to On if the port is used for incoming Gateway calls.

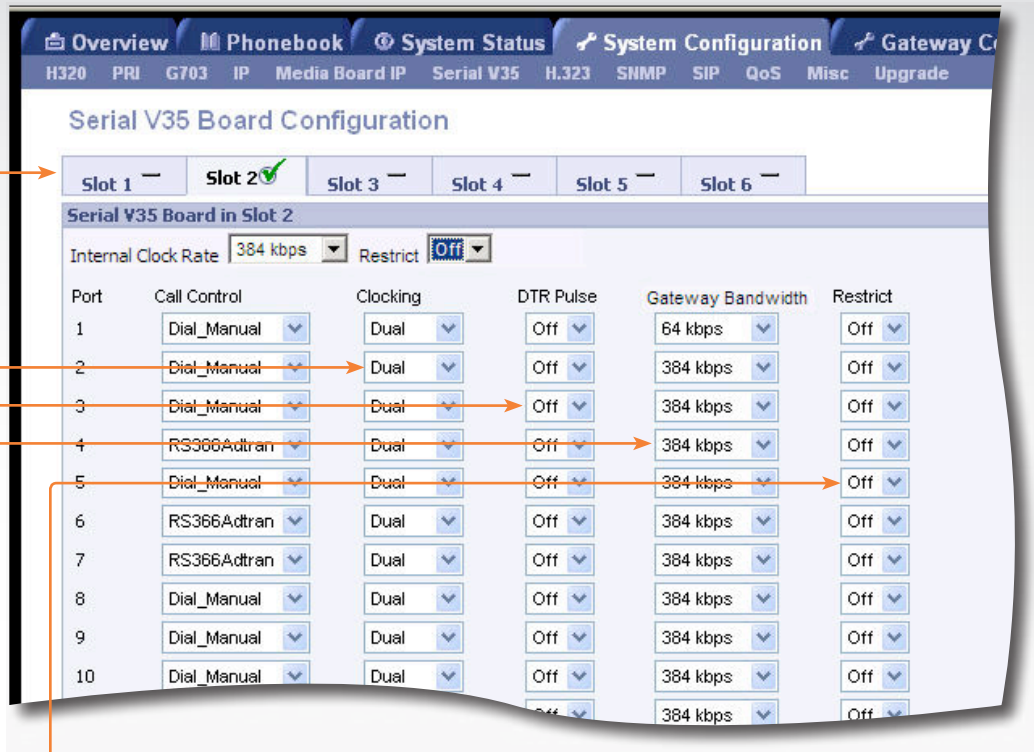
Off: Set to Off if the port is not used for incoming Gateway calls. (Default: Off)

Gateway Bandwidth, port 1-32

The incoming connection Bandwidth may be set to:

64 kbps, 128 kbps, 192 kbps, 256 kbps, 320 kbps, 384 kbps, 512 kbps, 768 kbps, 1152 kbps, 1472 kbps, 1536 kbps and 1920 kbps (Default: 384 kbps).

This is for incoming Gateway calls and has no effect when used for MCU calls.



Save - Press the Save button to save changes.

Restrict, port 1-32

The port using internal clocking must match the Restrict-settings of the Internal Clock Rate.

On: When set to On, call rates increments in steps of 56 kbps.

Off: When set to Off, call rates increments in steps of 64 kbps. (Default: Off)

H.323 Configuration

H.323 Configuration - Gatekeeper Settings for Net 1 and Net 2

When dialling directly in to a conference from H.323, this requires the use of H.323 numbers. To achieve this the TANDBERG MPS must be registered to a Gatekeeper.

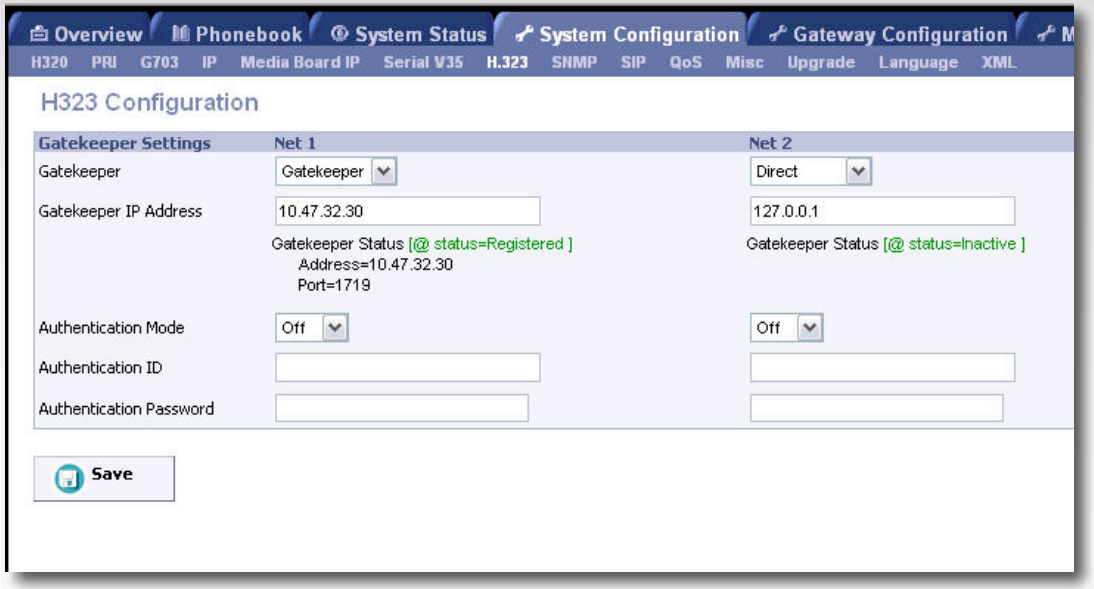
Gatekeeper: Set Gatekeeper to Gatekeeper to enable the MPS to register to a Gatekeeper.

- The Gatekeeper IP Address must also be filled in.
- When registered, the H.323 Gatekeeper Status will show: Registered, Gatekeeper's IP address and the Port used
- Problems with registration will be shown as: Registering. A Red alarm-symbol will be shown on the Conference Overview page.

Direct: Set Gatekeeper to Direct if the MCU should not register to any Gatekeeper.

- The H.323 Gatekeeper Status will show: Inactive.

The Gatekeeper settings can be configured differently for Network #1 and Network #2.



Gatekeeper IP Address

Enter the Gatekeeper IP Address that the MCU should register with.

Gatekeeper Status

H.323 Gatekeeper Status shows current status of Gatekeeper registration.

Save - Press the Save button to save changes.

Authentication Mode

The Authentication Mode is set to Auto or Off in order to signal that the system supports Gatekeeper authentication or not.

The NTP must be configured on the IP configuration page if Authentication is to be used.

Authentication ID / Authentication Password

The Authentication ID and the Authentication Password are used by a Gatekeeper to authenticate the system.

The system can be authenticated by one or more Gatekeepers.

SNMP Configuration

SNMP Configuration

SNMP is used for monitoring and configuring different units in a network. The SNMP Agent responds to requests from SNMP Managers (a PC program etc.). SNMP traps are generated by the agent to inform the manager about important events.

SNMP - Simple Network Management Protocol

MIB - Management Information Base

SNMP Mode

On: Set the SNMP Mode to On to enable the SNMP Agent. This will enable generating of SNMP traps, and the ability to Read and Write to the System MIB.

Off: Set the SNMP Mode to Off to disable the SNMP Agent. This will disable generating of SNMP traps, and the MIB entries can neither be read nor written to.

Read Only: When SNMP Mode is Read Only, SNMP traps are generated, and the system MIB can only be read, not written to.

Traps Only: When SNMP Mode is Traps Only, SNMP traps are generated, but the system MIB cannot be read or written to.

Community Name

SNMP Community names are used to authenticate SNMP requests. SNMP requests must have this 'password' in order to receive a response from the SNMP agent in the MCU.

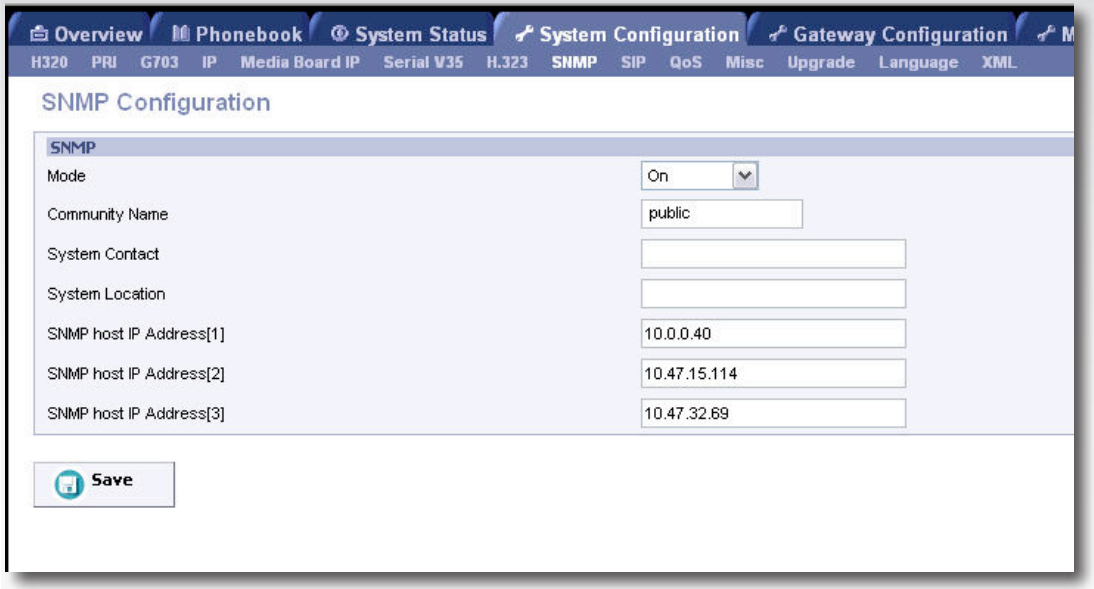
NOTE! The SNMP Community name is case sensitive.

System Contact

Used to identify the system contact via SNMP tools such as TANDBERG Management Suite or HPOpenView.

System Location

Used to identify the system location via SNMP tools such as TANDBERG Management Suite or HPOpenView.



Save - Press the Save button to save changes.

SNMP Host IP Address (1-3)

Identifies the IP-address of the SNMP manager.

- Up to three different SNMP Trap Hosts can be defined.
- Your LAN administrator should provide the correct values for these fields

SIP Configuration

SIP Mode

On: Set the SIP mode to On to enable the system for incoming and outgoing SIP calls.

Off: Set the SIP mode to Off to disable incoming and outgoing SIP calls from the system

Server Address

The Server Address is the manually configured address for the outbound proxy and registrar. It is possible to use a fully qualified domain name, or an IP address. The default port is 5060 for TCP and UDP, but another one can be provided.

Examples:

```
sipserver.example.com
sipserver.example.com:5060
10.0.0.2
10.0.0.2:5060
```

Server Type

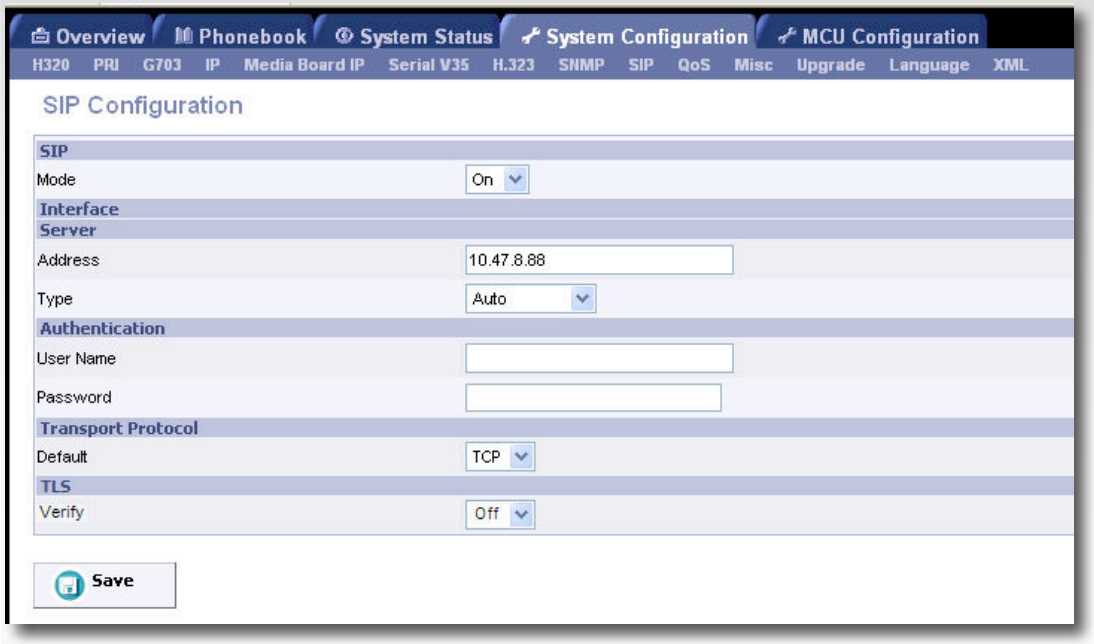
Set the correct SIP server type to let the MPS communicate with the SIP server. The following selections are available:

Auto, Nortel, Microsoft, Cisco, Alcatel, Experimental.

Authentication User Name / Authentication Password

This is the user name part and password part of the credentials used to authenticate toward the SIP Server.

NOTE: The DNS must be configured on the IP configuration page when using SIP



Save - Press the Save button to save changes.

Default Transport Protocol

Sets the default transport protocol towards the SIP server.

TCP: Set TCP as the default transport protocol.

UDP: Set UDP as the default transport protocol.

The most convenient transport protocol differs from server to server. i.e. when Server Type is set to “Nortel”, the transport default should be UDP, when set to “Microsoft LCS server” it should be TCP.

SIP Verify TL TLS

For TLS connections a CA-list can be uploaded from the web interface.

On: Set to On to verify TLS connections. Only TLS connections to servers, whom x.509 certificate is validated against the CA-list, will be allowed.

Off: Set to Off to allow TLS connections without verifying them. The TLS connections are allowed to be set up without verifying the x.509 certificate received from the server against the local CA-list. This should typically be selected if no SIP CA-list has been uploaded.

TLS (Transport Layer Security) - is a transport protocol used over LAN.
CA - Certificate authority, issuer of (root) certificates.

Current RFC’s and Drafts Supported

Read about the Current RFC’s and Drafts Supported for SIP in the Appendices section.

QoS (Quality of Service) Configuration

QoS Configuration (network 1-2)

The network must support Quality of Service (QoS) for these settings to work. You can configure the Quality of Service mode differently for Network #1 and Network #2.

QoS Mode (network 1-2)

Precedence: Select IP Precedence QoS method and see QoS Mode Configuration for details.

Diffserv: Select Diffserv QoS method and see QoS Mode Configuration for details.

Off: No QoS is used.

QoS Mode Configuration

- Diffserv is used to define which priority audio, video, data and signalling packets should have in an IP network. The priority ranges from 0 to 63 for each type of packets.
- Precedence is used to define which priority audio, video, data and signalling should have in an IP network. The higher the number, the higher the priority. The priority ranges from 0 (Off) to 7 for each type of packets.
- In addition to Precedence, Type of Service can be used and enables the user to define what type of connection that should be chosen for the IP traffic. This helps a router select a routing path when multiple paths are available.

QoS Type of Service Priority

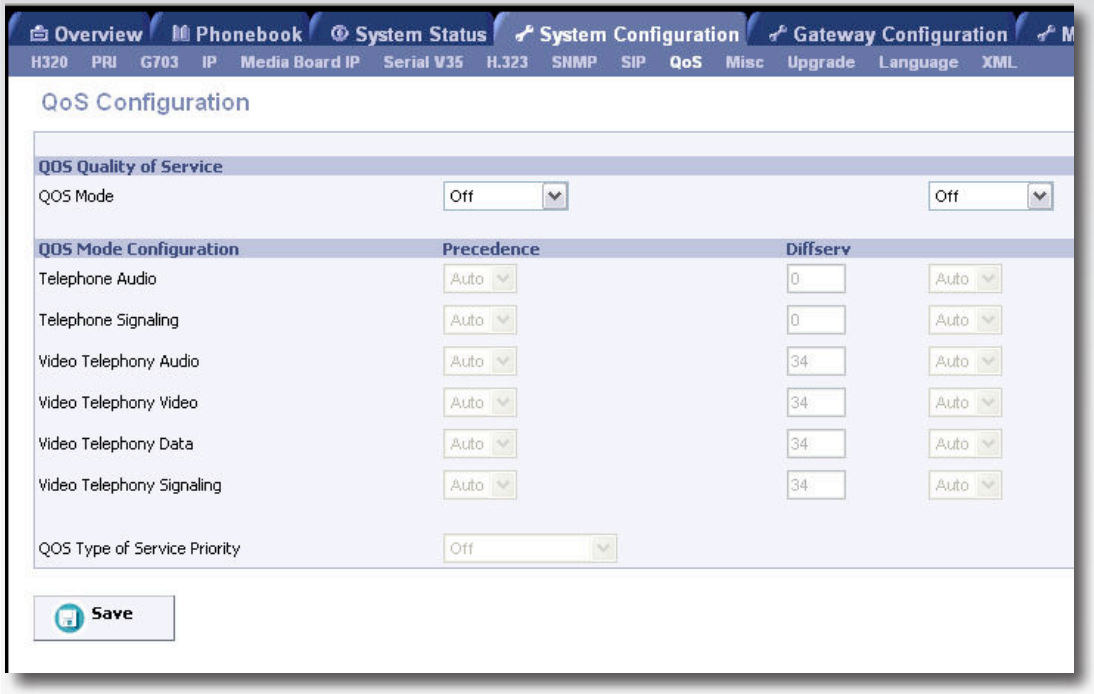
Minimum Delay: Will choose a route where minimum delay is prioritized.

Maximum Throughput: Will choose the route with highest bandwidth is prioritized.

Maximum Reliability: Will choose the route where minimum packet loss is prioritized.

Minimum Cost: Will choose the cheapest connection available.

Off: Quality of Service not active.



Save - Press the Save button to save changes.

Miscellaneous Configuration

Configuration - System name

To change the System Name of the TANDBERG MPS, enter the new system name in the <System Name> field.

Services

The IP services can be independently disabled to prevent access to the TANDBERG MPS.

HTTP

On: Set HTTP to On to enable HTTP Service.

Off: Set HTTP to Off to disable HTTP Service.

HTTPS

On: Set HTTPS to On to enable HTTPS Service.

Off: Set HTTPS to Off to disable HTTPS Service.

Telnet

On: Set Telnet to On to enable Telnet Service.

Off: Set Telnet to Off to disable Telnet Service.

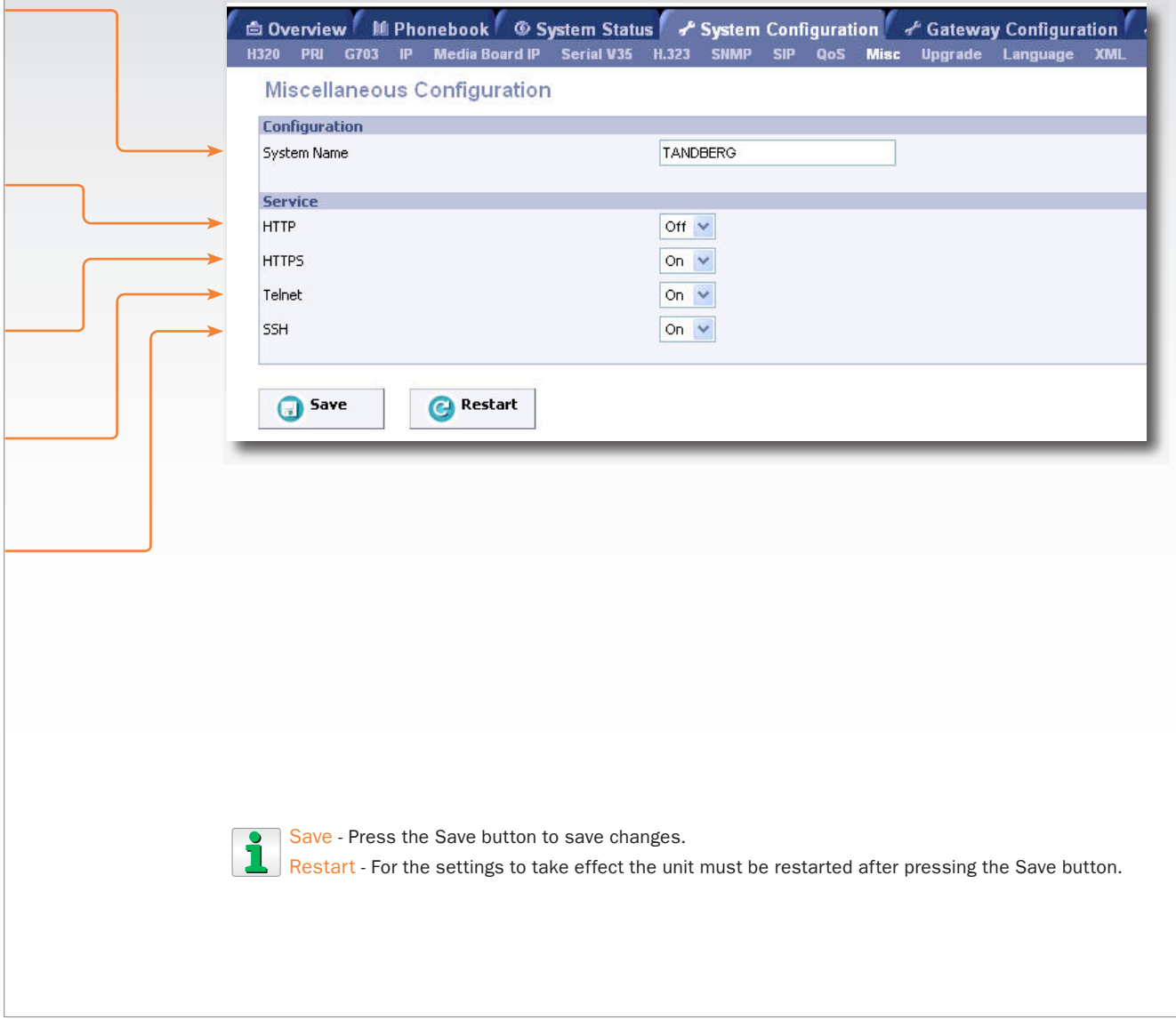
SSH (Secure Shell)

On: Set SSH to On to enable SSH Service.

Off: Set SSH to Off to disable SSH Service.

Change Password

To change the password of the system, you need to log into the Command Line Interface. Read about the API commands in the TANDBERG MPS API Guide. Go to: http://www.tandberg.com/docs.



Save - Press the Save button to save changes.

Restart - For the settings to take effect the unit must be restarted after pressing the Save button.

Software Upgrade

Software Upgrade

New software to the TANDBERG MPS can be installed via the Software Upgrade web page.

The Software Upgrade page shows the current software version, the hardware serial number and the installed options and the option keys.

Important - Before Upgrading

To upgrade the TANDBERG MPS, a valid Release Key and Software file is required. Please contact your TANDBERG representative for more info.

System Information - Software Version: Shows the currently installed Software version.

System Information - Hardware Serial Number: Shows the unique hardware serial number of the TANDBERG MPS at hand. The TANDBERG MPS serial number must be provided when ordering a Software Upgrade.

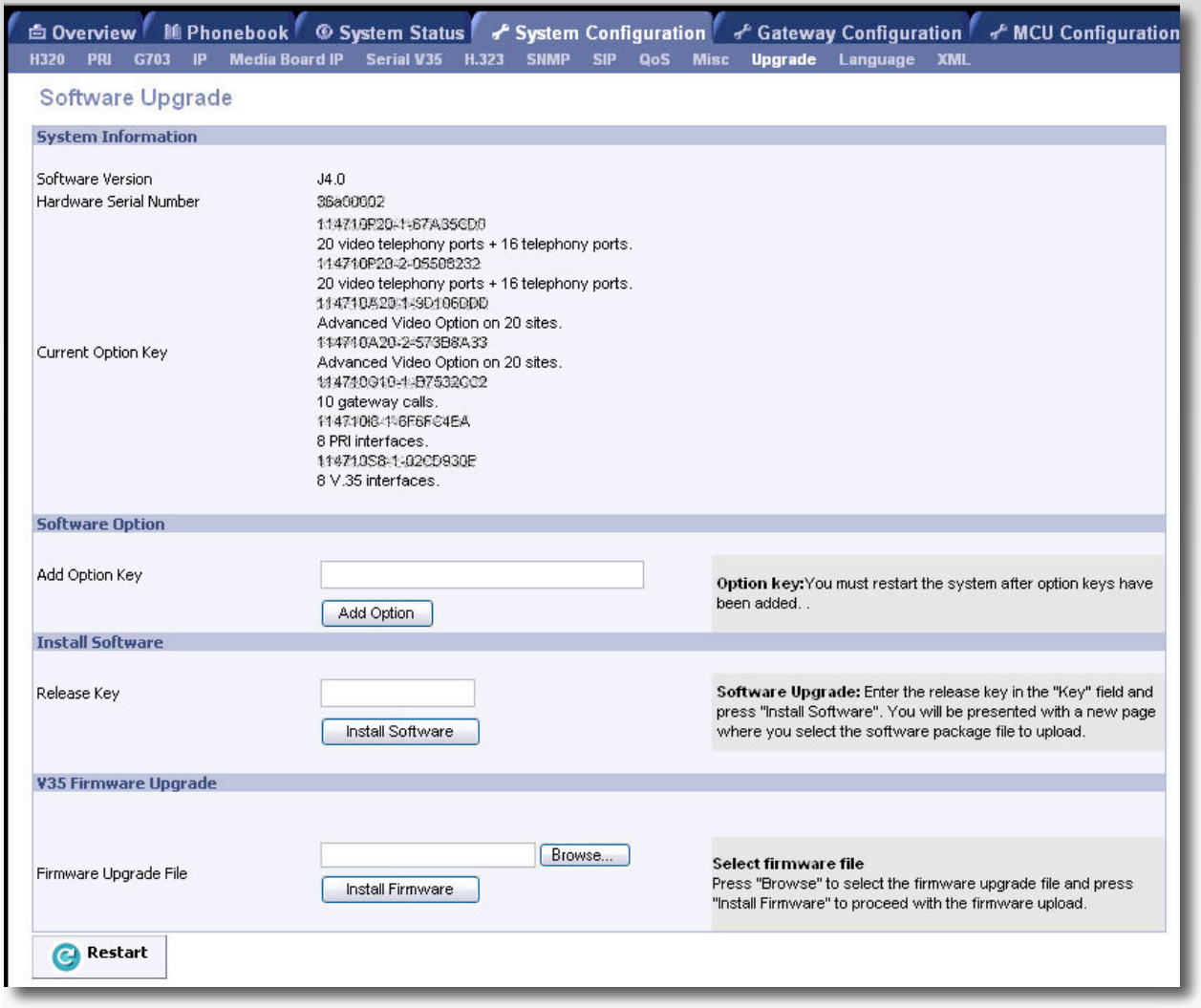
System Information - Current Option Key: Shows the current installed Option Keys.

Software Option - Add Option Key: Enter the option key in the Key field and press 'Add Option'. The system will validate the key. When validated, feel free to add more option keys in the same manner. However, to get the new options up and running, the TANDBERG MPS must be restarted. Please use the Restart button on the bottom of this page.

Install Software - Release Key: Enter the release key in the Key field and press 'Install Software'. You will be presented with a new page where you select the software package file to upload.

V.35 Firmware Upgrade - Firmware Upgrade File: Press 'Browse' to select the firmware upgrade file and press "Install Firmware" to proceed with the firmware upload.

There are two different version of the TANDBERG MPS software file. One standard software file (`s41000jxx.tar.gz`), and one software file (`s41001jxx.tar.gz`) without Encryption (without AES and DES support).



For changes to take effect the unit must be restarted.

Restart - Press the Restart button. This will restart the MPS.

Language, Set Language and Language files

Set Language

The Set Language allows you to view and change the language in the web interface and dialogs.

Select Web Interface Language

Select the desired Web Interface Language and press Set to activate.

Select Dialog Language

Select the desired Dialog Language and press Set to activate.

File Management

The File Management allows viewing and changing pictures, sounds and symbols, which are shown to the participants when connecting to, and during a conference on the MPS.

Different files can be uploaded for different languages. If you want to use files that are language independent, see File Management in the MCU Configuration section.

Manage Files for Language

Select the desired language for managing language dependent files only. Press the Manage button to activate.

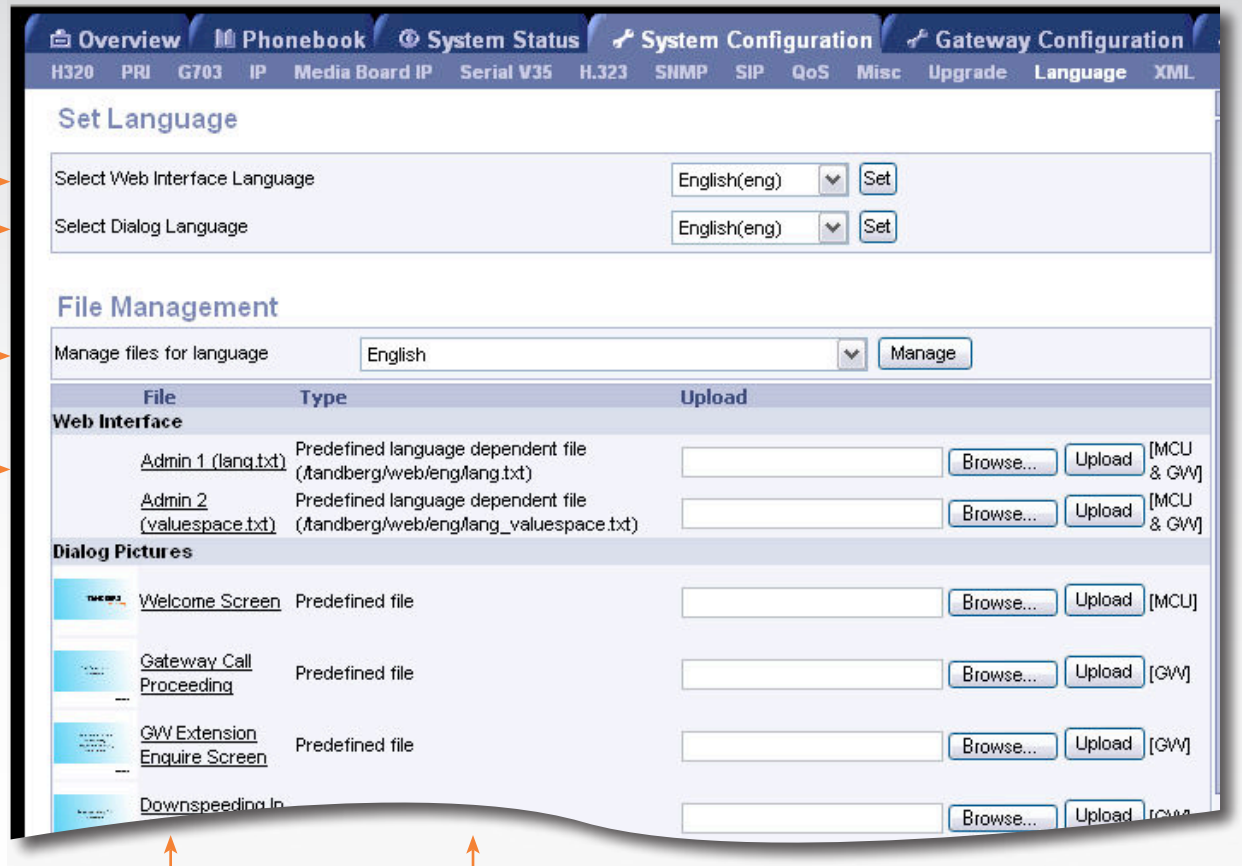
Web Interface

The following web interface options can be specified:

- Language - Upload language file (MCU & GW).
- Value Space - Upload Value Space file (MCU & GW).

Both files should be uploaded for a given language.

To add a new file, press Browse to find the file, and then press Upload. For each customized file, a Delete button will be added in the Type column.



File

Press the name of the File on the left side of the window to view or hear the content of the file.

Type

- When a customized file has been uploaded the text “Predefined file“ is replaced with a Delete button.
- When a customized file has been deleted the Delete button is replaced with the text “Predefined file“.

Browse

Press the Browse button to find a file.

Upload

Press the Upload button to upload the file.

Language, Files.., Dialog pictures

File Management

The File Management allows viewing and changing pictures, sounds and symbols, which are shown to the participants when connecting to, and during a conference on the MPS.

Different files can be uploaded for different languages. If you want to use files that are language independent, see File Management in the MCU Configuration section.

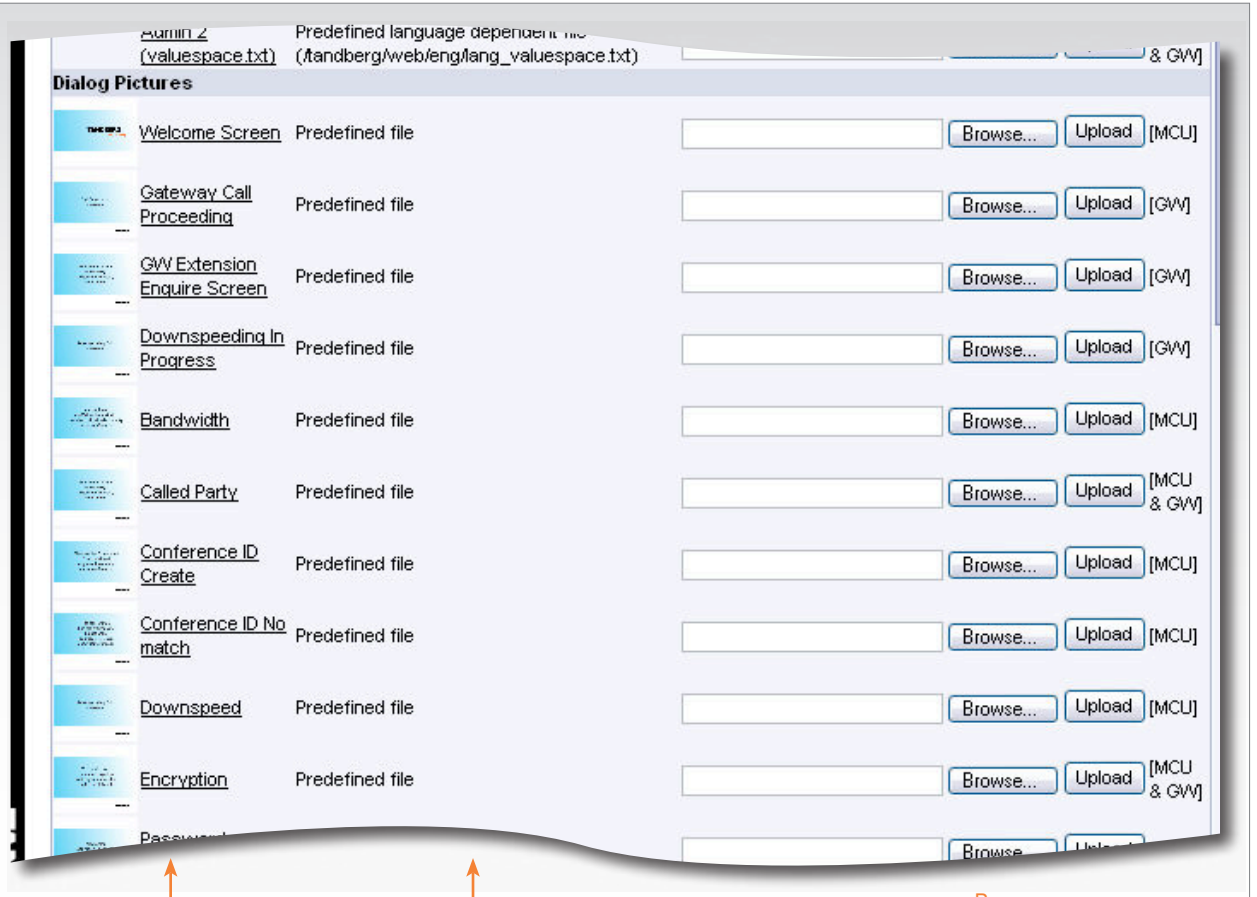
Dialog Pictures

Examples of dialog pictures to be specified:

- Welcome Screen
- Gateway Call Proceeding
- GW Extension Enquire Screen
- Downspeeding In Progress
- Called Party
- Bandwidth
- Conference ID Create
- Conference ID Create No Match
- Downspeed
- Encryption
- Password Activate
- Password Create
- Password Enter
- Password Reject
- Ambigous Conference ID
- Now, Enter Conference ID
- Not Started Activate
- Not Started Wait
- Only Participant

To add a new file, press Browse to find the file, and then press Upload. For each of the customized files, a Delete button will be added in the Type column.

Allowed picture format: JPEG (.jpg) files that are not gray scale and non-progressive coded.
Recommended maximum size is 352x288.



File

Press the name of the File on the left side of the window to view or hear the content of the file.

Type

- When a customized file has been uploaded the text “Predefined file“ is replaced with a Delete button.
- When a customized file has been deleted the Delete button is replaced with the text “Predefined file“.

Browse

Press the Browse button to find a file.

Upload

Press the Upload button to upload the file.

Language, Files.., Dialog Sounds and Symbols

File Management

The File Management allows viewing and changing pictures, sounds and symbols, which are shown to the participants when connecting to, and during a conference on the MPS.

Different files can be uploaded for different languages. If you want to use files that are language independent, see File Management in the MCU Configuration section.

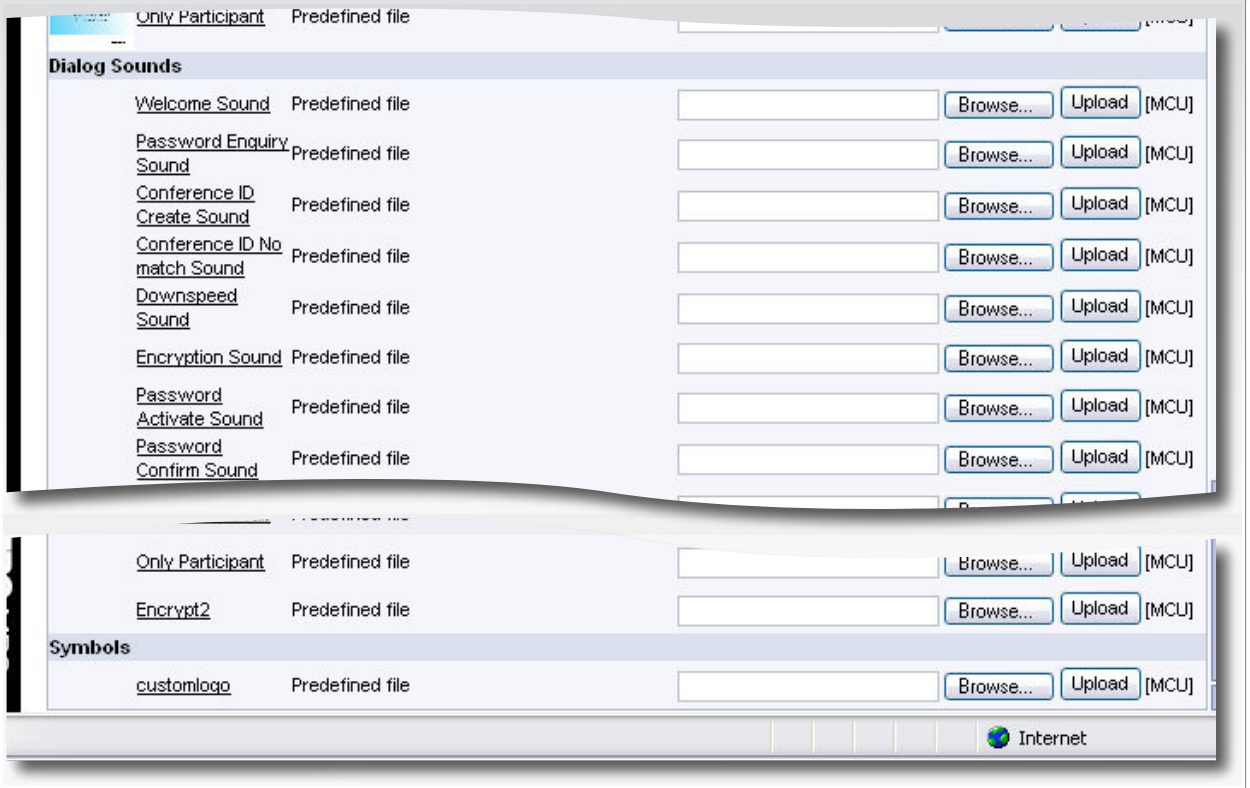
Dialog Sounds

Examples of dialog sound to be specified:

- Welcome Sound
- Password Enquiry Sound
- Conference ID Create Sound
- Conference ID Create No Match Sound
- Downspeed Sound
- Encryption Sound
- Password Activate Sound
- Password Confirm Sound
- Password Create Sound
- Password Wrong Sound
- Password Activate Please Sound
- Ambigous Conference ID Sound
- Now, Enter Conference ID Sound
- Call Proceeding
- Extension Enquire Sound
- Not Started Activate
- Not Started Wait
- Only Participant
- Encrypt2

To add a new file, press Browse to find the file, and then press Upload. For each of the customized files, a Delete button will be added in the Type column.

Allowed sound format: 16bit 8kHz mono Wave (.wav) file.



Symbols

A Custom Logo can be specified:

- Recommended size is 64 x 64 pixels
- The MPS must be restarted to display the Custom Logo
- The Custom Logo will be presented in the upper left corners on all conferences hosted by the MPS. This logo can be used to identify/verify the MPS hosting the conference.

The logo will not be present in conferences which have enabled and are using optimal Voice Switch, oVS.

To add a new file, press Browse to find the file, and then press Upload. For each of the customized files, a Delete button will be added in the Type column.

Configuration using XML

XML Document

The XML Uploading page, allows administrators to upload multiple configuration changes to the MPS instead of manually setting each entry through the web interface.

By pasting in a valid xml file, users may update the MPS Directory and Configuration. It is also possible to execute commands by pasting in valid command xml.

Max size approx. 150 lines

For longer XML strings, use a normal HTTP POST towards http://MPS-IP/putxml with the body being the raw XML document.

The header “Content-Type” must be set to “text/xml” for correct processing.

The required XML format for Directories, Configuration and Commands can be found at the following locations:

`http://<MPS IP Address>/command.xml`

`http://<MPS IP Address>/directory.xml`

`http://<MPS IP Address>/configuration.xml`

If the MPS is configured for HTTPS:

`https://<MPS IP Address>/command.xml`

`https://<MPS IP Address>/directory.xml`

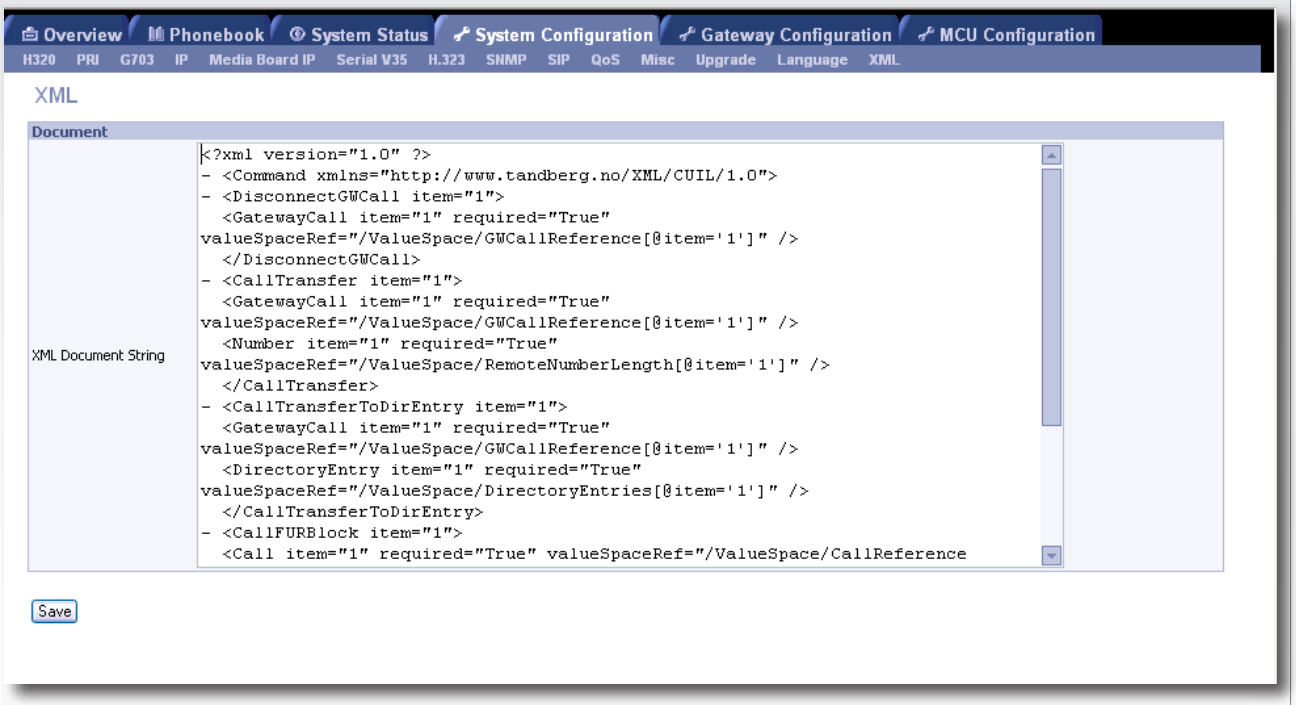
`https://<MPS IP Address>/configuration.xml`

Using the XML Document

1. Open the XML file, for example command.xml
2. Copy text from the XML file
3. Paste text into the XML Document editor field
4. Make the desired changes
5. Press the Save button to activate the changes

Some configuration changes may require a restart of the MPS to take effect.

Example: Text is copied from the command.xml file and pasted into the XML Document editor field.



Save - Press the Save button to save changes.

Certificate Management

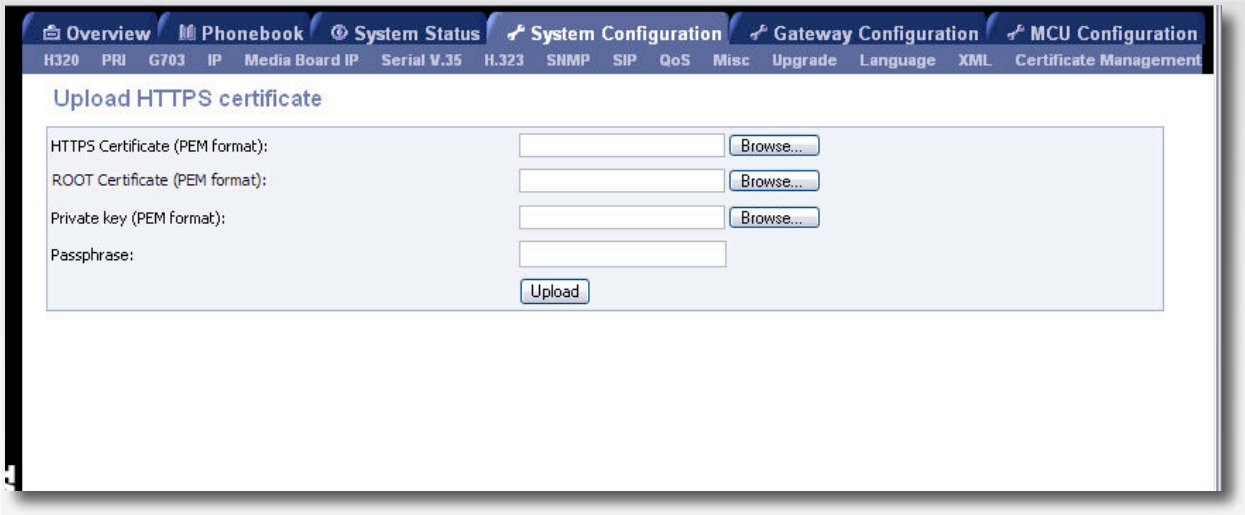
Manage Certificates

Allows an administrator to upload X.509 certificates for use with HTTPS. These certificates can be created from any trusted CA (Certificate Authority) to prevent possible security warnings in the web browser due to the default self signed certificate.

To install an HTTPS certificate, the following is needed:

- The HTTPS certificate (.PEM format)
- ROOT Certificate (.PEM format, optional)
- The Private key (.PEM format)
- The Passphrase (optional)

NOTE - Although a Passphrase is optional; if the certificate was created using a passphrase then the passphrase must be entered to make use of the certificate.



Browse - Press the Browse button to locate the HTTPS certificate and Private key.

Upload - Press the Upload button to upload the HTTPS certificate, Private key and Passphrase (if used).

NOTE - The MPS must be restarted for the new certificate to take affect.

Chapter 6

Gateway configuration

The Gateway Configuration section contains information about how to configure and use the Gateway on the TANDBERG MPS. Reading this section makes you familiarize yourself with the configuration menu and functions important for the correct configuration and functioning of the Gateway. You will also find examples of setting up Dialling Rules on the Gateway with different types of network configuration.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

In this chapter...

Gateway Functionality and Dialling Rules

How to enable Gateway Funtionality

In order to enable the Gateway functionality in the MPS, the Gateway Software option must be set.

Gateway Software Option

The software option key is set in the web interface, see the System Configuration > Upgrade section.

About the Option Key

An option key for the Gateway can be obtained through the regular TANDBERG channels. The option key states how many Gateway sessions the MPS is able to run in parallel.

Gateway Sessions

A Gateway session requires 2 ports (one for the source side and one for the destination side), thus making 80 the maximum number of Gateway sessions for the MPS 800. For MPS 200 the maximum is 20 sessions.

Gateway Calls Overview

Once the correct option key for the Gateway is entered (and the system restarted), the web interface will include a Gateway Calls Overview page, see the Overview > Gateway Calls Overview section.

Gateway Configuration

Once the correct option key for the Gateway is entered (and the system restarted), an extra tab for Gateway Configuration will also be added to the web interface.

For further information about endpoint support of TCS-4 see the MXP Administrator's Guide on http://www.tandberg.com/docs.

Dialling Rules

The dialling rules must be set in order for the MPS to handle different types of Gateway calls. The different Gateway call types supported by the MPS are described below.

DID

Direct Inward Dialling (DID) will provide you with direct Dial In numbers for your endpoints.

DID will do a direct mapping between your ISDN number and the H.323 E.164 Alias. If you have assigned a range of ISDN numbers to your ISDN PRI line, each ISDN number will map to a single IP endpoint.

We recommend that if more than one PRI line is used all PRI lines should have a common number range, see Trunk Groups in the PRI Board Configuration section.

Example with DID:

1. A DID dialling rule exists that maps the PRI number range from 67124000 to 67124050 to the H.323 E.164 Alias range 94000 - 94050.
2. To call an IP endpoint with H.323 E.164 Alias 94020 from ISDN, dial the ISDN number 67124020. • The Gateway starts the call to the IP endpoint and the "Call proceeding" picture and sound are initiated to your endpoint.
3. When the call is connected audio and video are transmitted through the Gateway.

IVR Services

Interactive Voice Response (IVR), also called extension Dial In, provides you with a single Dial In number. The caller uses telephone tones (DTMF) to enter the extension address of the endpoint to be called. It is an automated answering system that directs the call to the endpoint indicated by the caller.

IVR is useful when you have limited PRI numbers on your PRI line.

Example with IVR:

1. A video conferencing system calls into the Extension Dial In number of an IVR service.
2. The Gateway activates the 'Welcome' picture and sound.
3. The user of a video conferencing system enters the extension (H.323 E.164 Alias) followed by the # (pound-sign).
4. The Gateway starts to call the IP endpoint and the "Call proceeding" picture and sound are activated.
5. When the call is connected, the audio and video are transmitted through the Gateway.

IVR + TCS-4

This call type is the same as described in IVR but extended by an additional method, the TCS-4, to signal the extension address to the Gateway. In this mode the extension number can be indicated with IVR or with TCS-4 signalling.

TCS-4 allows an H.320 based videoconferencing endpoint (ISDN, V.35, V.35RS366 and G.703) to dial an IP endpoint directly, without having to (manually) enter the extension number by DTMF. The endpoint will send the extension number as a TCS-4 signal to the Gateway. If no TCS-4 extension is sent from the endpoint, then IVR will be used.

To use TCS-4, it needs to be supported by the video conferencing system. .

Example with IVR + TCS-4:

1. A video conferencing system calls into the Extension Dial In number of an IVR + TCS-4 service, using <Dial In number of this service>*<extension number>.
2. The Gateway starts to call the IP endpoint and the "Call proceeding" picture and sound are activated.
3. When the call is connected audio and video are transmitted through the Gateway.
4. TANDBERG endpoints allow storing the complete dial string in the Phone Book to automate dial through.

Gateway Configuration

Dialling Rules

Add new Dialling Rule

1. Configure the new Dialling Rule from the row on top
2. Press the Add New button to activate the dialling rule
3. The new rule is added as the Rule #1 in the list

Modify an existing Dialling Rule

1. Locate the Dialling Rule to be modified (1, 2, 3, etc.)
2. Modify the Dialling Rule and press the Save button to activate the modified Dialling Rule

Delete a Dialling Rule

1. Locate the Dialling Rule to be deleted (1, 2, 3, etc.)
2. Press the Delete button

Order of Dialling Rules

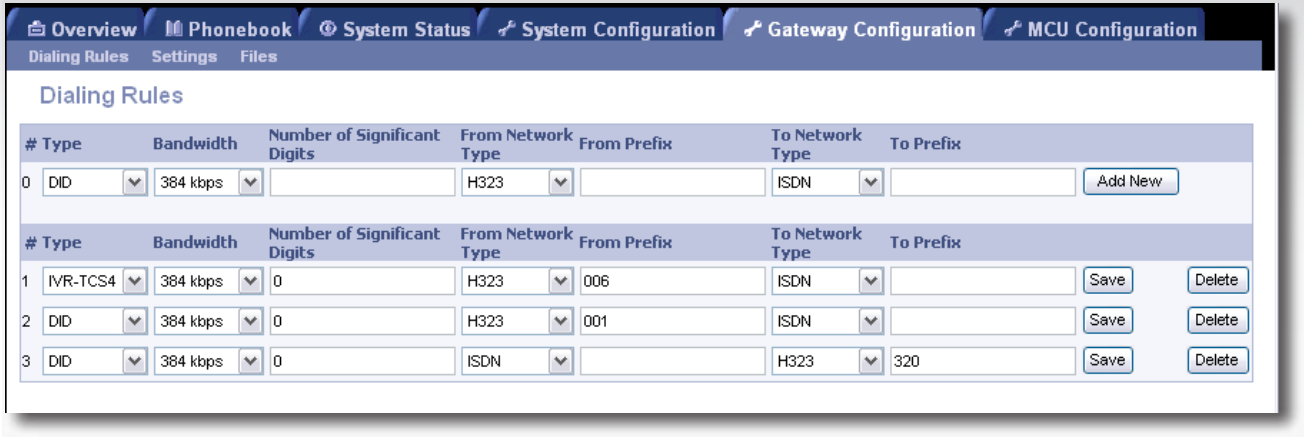
The rules are processed in the order they appear in the list. Rule #1 is first and Rule #2 next, etc. In cases where the order of rules matters, make sure they are added in the right order. A new rule is added on top as Rule #1.

Number of Dialling Rules

There is no limitation in the MPS as to how many Dialling rules you want to set up.

However, bear in mind that the total number of registrations on the Gatekeeper can be limited.

For further information about endpoint support of TCS-4 see the MXP Administrator's Guide on http://www.tandberg.com/docs.



Type

DID - Direct Inward Dialling

IVR - Interactive Voice Response

IVR-TCS4 - IVR and TCS-4 Service

Bandwidth

This value sets the maximum call rate for the given Gateway service: Telephone, 64 kbps, 128 kbps, 192 kbps, 256 kbps, 384 kbps, 512 kbps, 768 kbps, 1152 kbps, 1472 kbps, 1920 kbps.

Number of Significant Digits

Specify the number of significant digits of the incoming number that should be used to:

- Do a rule matching
- Generate an outgoing number

Setting this field to 0 (zero) will make the Gateway check the complete incoming number for rule matching.

From Network Type - To Network Type

From Network Type - The call type on for incoming calls. Supported call types: H.323, ISDN, V.35, V.35RS366 and G.703 (leased line).

From Prefix - This setting instructs what to match on the incoming call. The significant number of digits is applied on the input, before matching From Prefixes.

To Network Type - The call type on the destination side. Supported call types: H.323, ISDN, V.35, V.35RS366 and G.703 (leased line).

To Prefix - The To Prefix is used to append a number to the matched number (now without the From Prefix). The To Prefix will be first in the outgoing dialled number from the Gateway.

Add New

Adds a new dialling rule to the list below and activates the new dialling rule.

Take care! Pressing the Add New button will create a new Dialling Rule at the top of the list. In cases where the order of the Dialling Rules matters, this may affect those calls, see Example #1 on next page.

Save

Press the Save button to save changes to a Dialling Rule.

Delete

Press the Delete button to delete a Dialling Rule.

Take care! No warning will appear.

Gateway Configuration, *cont...*

Examples of dialling rules with ISDN Gateway

Example with DID

Your ISDN PRI number range for the Gateway is from 67124000 to 67124050.

Your IP endpoints have the numbers (H.323 E.164 Alias) in the range 4000 - 4050.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to 8
4. Set From Network Type to ISDN
5. Set From Prefix to 6712
6. Set To Network Type to H.323
7. Leave To Prefix empty

To call an IP endpoint with H.323 E.164 Alias 4020 from ISDN, dial the ISDN number 67124020.

The Order of Dialling Rule Matters

To be able to do “empty” incoming prefix matching as the example above, this Dialling Rule must be the first that is added to the mapping rules database. If not, this rule will match with any other prior added dial in rules.

If you want your IP endpoints to have numbers in the range 94000 – 94050

- Set To Prefix to 9 instead

Example with IVR-TCS4

Your ISDN PRI number for Gateway usage is 67124000.

Add Dialling Rule

1. Set Type to IVR-TCS4
2. Select Bandwidth
3. Set Number of Significant Digits to 8
4. Set From Network Type to ISDN
5. Set From Prefix to 67124000
6. Set To Network Type to H.323
7. Leave To Prefix empty

To call an IP endpoint with H.323 E.164 Alias 4020 from ISDN, dial the ISDN number 67124000*4020.

The notation to separate the ISDN number from the extension by ‘*’ is only available on certain types of endpoints. Consult the endpoint documentation to find out if and how TCS-4 is supported.

Using IVR instead

If you want to use IVR instead dial 67124000 and enter ‘4020#’ using DTMF.

H.323 to ISDN Gateway

It is also possible to configure an H.323 to ISDN Gateway.

Example with DID:

The GW shall be configured to have an H.323 prefix 0047.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to 0
4. Set From Network Type to H.323
5. Set From Prefix to 0047
6. Set To Network Type to ISDN
7. Leave To Prefix empty

To call an ISDN endpoint with ISDN number 67141234 from H.323, dial the number 004767141234.

Setting the Gateway’s Dial In number to 0047 will make the Gatekeeper route all H.323 calls to an E.164 alias starting with 0047 to this Gateway.

- This means that all other equipment has to be configured with an E.164 alias that doesn’t start with 0047.
- Exceptions to this rule are other Gateways to perform load balancing.

Example with IVR:

The GW shall be configured to have an H.323 Dial In number 1.

Add Dialling Rule

1. Set Type to IVR
2. Select Bandwidth
3. Set Number of Significant Digits to 0
4. Set From Network Type to H.323
5. Set From Prefix to 1
6. Set To Network Type to ISDN
7. Leave To Prefix empty

To call an ISDN endpoint with ISDN number 67141234 from H.323, dial 1 and enter ‘67141234#’ using DTMF.

Dialling rules for telephone connections are set up accordingly, with Bandwidth set to Telephone.

For further information about endpoint support of TCS-4 see the MXP Administrator’s Guide on http://www.tandberg.com/docs.

The J3 software and above supports ISDN to H.323 Gateway.

Gateway Configuration, *cont...*

Examples of dialling rules with V.35 Gateway

V.35 Ports

To address the V.35 ports in a dialling rule, the physical V.35 ports have a numeric representation with two digits for each the interface number and module number.

Make sure that this string always contains 4 digits.

Range will therefore be 0101 to 0632 for corresponding V.35 cards/modules in slot 1 port 1 to slot 6 port 32.

Example:

Slot 1 & Port 1: 0101
Slot 1 & Port 2: 0102
Slot 1 & Port 3: 0103
....
Slot 1 & Port 30: 0130
Slot 1 & Port 31: 0131
Slot 1 & Port 32: 0132

Slot 6 & Port 1: 0601
Slot 6 & Port 2: 0602
Slot 6 & Port 3: 0603
....
Slot 6 & Port 30: 0630
Slot 6 & Port 31: 0631
Slot 6 & Port 32: 0632

For further information about endpoint support of TCS-4 see the MXP Administrator's Guide on http://www.tandberg.com/docs.

Example with DID:

You would like to have a DID number from H.323 to a V.35 port.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to empty or to 0 (zero)
4. Set From Network Type to H.323
5. Set From Prefix to 9
6. Set To Network Type to V.35
7. Leave To Prefix empty

To call a V.35 port 1 on module card 1, would require you to dial 90101 from the H.323 endpoint.

Configure Incoming V.35 Gateway

It is also possible to configure an incoming V.35 gateway.

- This can be setup as a V.35 DID or a V.35 IVR service.
- In the case of setting up an IVR service, make sure that your V.35 peripheral equipment supports DTMF relaying.

The J3 software and above supports H.323 to V.35 Gateway.

Example with DID (V.35 to H.323)

You would like to have a DID number from a V.35 port to a specific H.323 endpoint.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to empty
4. Set From Network Type to V.35
5. Set From Prefix to 0102
6. Set To Network Type to H.323
7. Set To Prefix to 55013

Make sure the Bandwidth selected matches the bandwidth of the V.35 port.

This would set up a Gateway call between the V.35 0102 port, to the H.323 endpoint, with alias 55013.

It is recommended not to use the same V.35 port for MCU calls and for Gateway calls because the support for incoming V.35 Gateway might conflict with the MCU V.35 setup.

The Gateway does only support H.320 to H.323 Gateway calls. There is no functionality to support Gateway routing internally within H.320 protocols (ISDN, V.35 and Leased Line).

It is recommended to set DTR pulse to On in the Serial V.35 configuration page for each port to be used for V.35 to H.323 Gateway calls.

Example with DID:

You would like to have a DID number from H.323 to a specific V.35 card.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to empty
4. Set From Network Type to H.323
5. Set From Prefix to 9
6. Set To Network Type to V.35
7. Set To Prefix to 01

To call V.35 port 4 on the already defined module card 1 would require you to dial 904 from the H.323 endpoint.

The To Prefix of 01 defines the use of Network Module 1 so this does not need to be included in the H.323 dial string.

Gateway Configuration, *cont...*

Examples of dialling rules with G.703 Gateway

G.703 Ports:

To address the G.703 ports in a dialling rule the physical G.703 ports have a numeric representation with two digits for each the interface number and the module number as well as a one digit representation for the call number.

Make sure that this string always contains 5 digits.

Range will therefore be 01011 to 06085 for corresponding G.703 cards/modules in slot 1 port 1 call 1 to slot 6 port 8 call 5.

Example:

Slot 1 Port 1 Call 1: 01011
Slot 1 Port 1 Call 2: 01012
Slot 1 Port 1 Call 3: 01013
....
Slot 1 Port 8 Call 3: 01083
Slot 1 Port 8 Call 4: 01084
Slot 1 Port 8 Call 5: 01085

Slot 6 Port 1 Call 1: 06011
Slot 6 Port 1 Call 2: 06012
Slot 6 Port 1 Call 3: 06013
....
Slot 6 Port 8 Call 3: 06083
Slot 6 Port 8 Call 4: 06084
Slot 6 Port 8 Call 5: 06085

Example with DID:

You would like to have a DID number from H.323 to G.703.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to empty or to 0 (zero)
4. Set From Network Type to H.323
5. Set From Prefix to 9
6. Set To Network Type to G.703
7. Leave To Prefix empty

To call G.703 call 1 on port 1 on module card 1 would require you to dial 901011 from the H.323 endpoint.

Example with DID:

You would like to have a DID number from H.323 to a specific G.703 card.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to empty or to 0 (zero)
4. Set From Network Type to H.323
5. Set From Prefix to 9
6. Set To Network Type to G.703
7. Set To Prefix to 01

To call G.703 call 3 on port 2 on module card 1 would require you to dial 9023 from the H.323 endpoint.

Example with DID:

You would like to have a DID number from H.323 to a specific G.703 call.

Add Dialling Rule

1. Set Type to DID
2. Select Bandwidth
3. Set Number of Significant Digits to empty or to 0 (zero)
4. Set From Network Type to H.323
5. Set From Prefix to 9
6. Set To Network Type to G.703
7. Set To Prefix to 01062

To call G.703 call 2 on port 6 on module card 1 would require you to dial 9 from the H.323 endpoint.

For further information about endpoint support of TCS-4 see the MXP Administrator's Guide on http://www.tandberg.com/docs.

The J3 software and above supports H.323 to G.703 Gateway.

Gateway Configuration, *cont...*

Gateway Settings

Interoperability

Experience has shown that some features - even though they are standardized - might cause interoperability problems with legacy video conferencing products.

By disabling features on this page, the Gateway can be used as a filter to ensure interoperability with legacy products.

When receiving a call with the Gateway configured with only one number in the PRI number range, all other incoming calls will be blocked until the ongoing call is established.

This is because the network will have no additional number available for the second call while setting up the call.

When the call is established the number will be available and any incoming call will connect as normal.

Natural Video

On: Set to On to allow video formats and capabilities such as interlaced video (iCIF/iSIF) to be transmitted through the Gateway.

Off: When set to Off no interlaced video capabilities will be transmitted.

Custom Video Formats

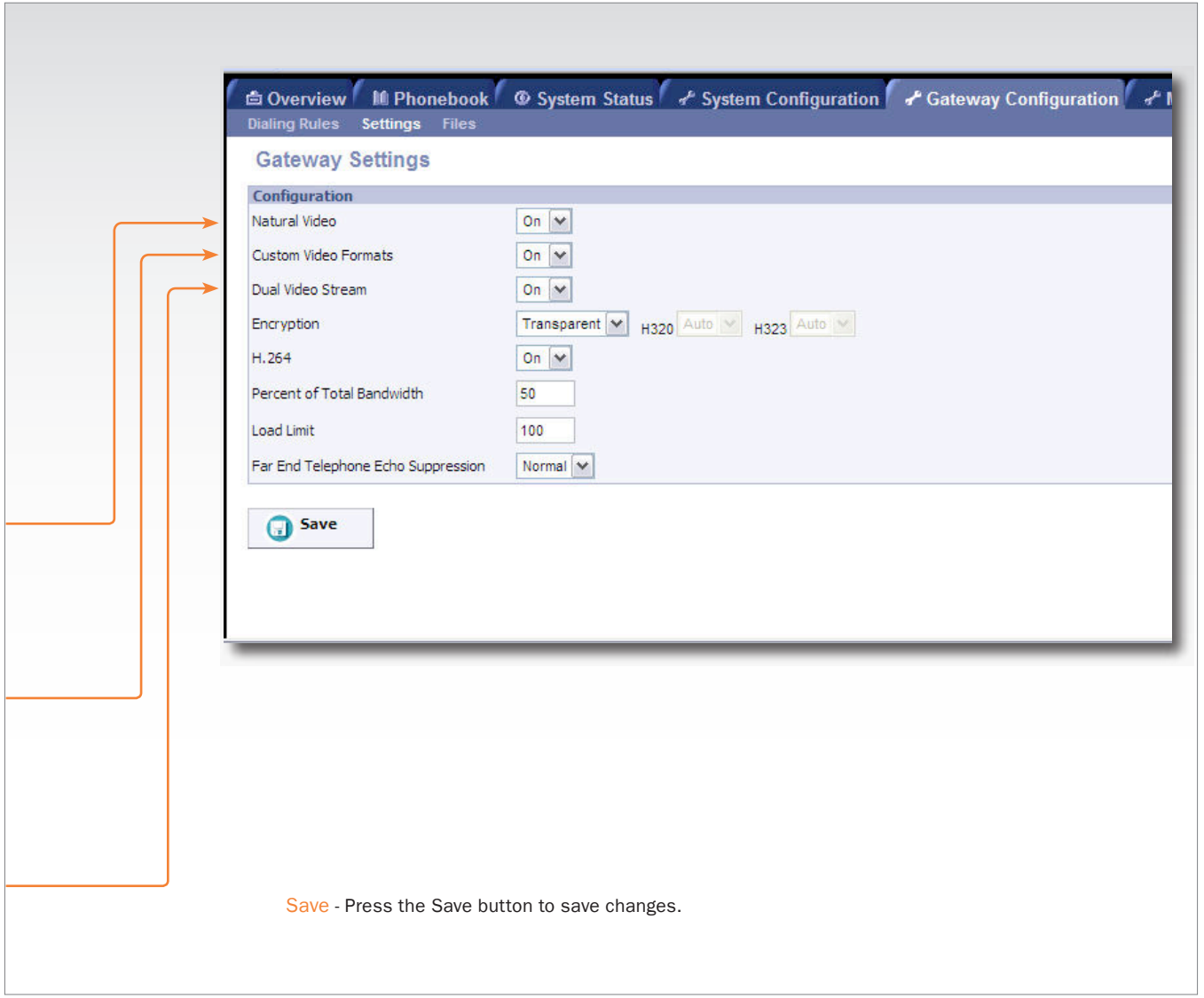
On: Set to On to allow H.263 custom video formats and capabilities such as SIF, 4SIF and VGA resolution to be transmitted through the gateway.

Off: When set to Off no H.263 custom video formats or capabilities will be transmitted.

Dual Video Stream (DuoVideo^{TF}/H.239)

On: Set to On to allows an additional video stream to be transmitted through the gateway using DuoVideo^{TF} or the H.239 protocols.

Off: When disabled no DuoVideo^{TF}, H.239 capabilities will be transmitted.



Save - Press the Save button to save changes.

Gateway Configuration, *cont...*

Gateway Settings

Interoperability

Experience has shown that some features - even though they are standardized - might cause interoperability problems with legacy video conferencing products.

By disabling features on this page, the Gateway can be used as a filter to ensure interoperability with legacy products.

Encryption

The Gateway will allow DES and AES 128 encrypted Gateway calls to take place. Encryption can operate in two modes.

Transparent: When set to Transparent, the Gateway will forward the encryption modes of the connected devices transparently. The encryption modes seen on each side of the Gateway are similar. The Gateway will only encrypt calls if both sides support encryption. The settings for H.320 and H.323 does not have any meaning in this mode.

Independent: When set to Independent, the Gateway will use the settings made for the H.320 (ISDN) and H.323 (IP) side. The encryption modes on each side of the Gateway are independent of each other. One endpoint might show that the call is encrypted while the other endpoint shows an unencrypted call.

H.320 and H.323

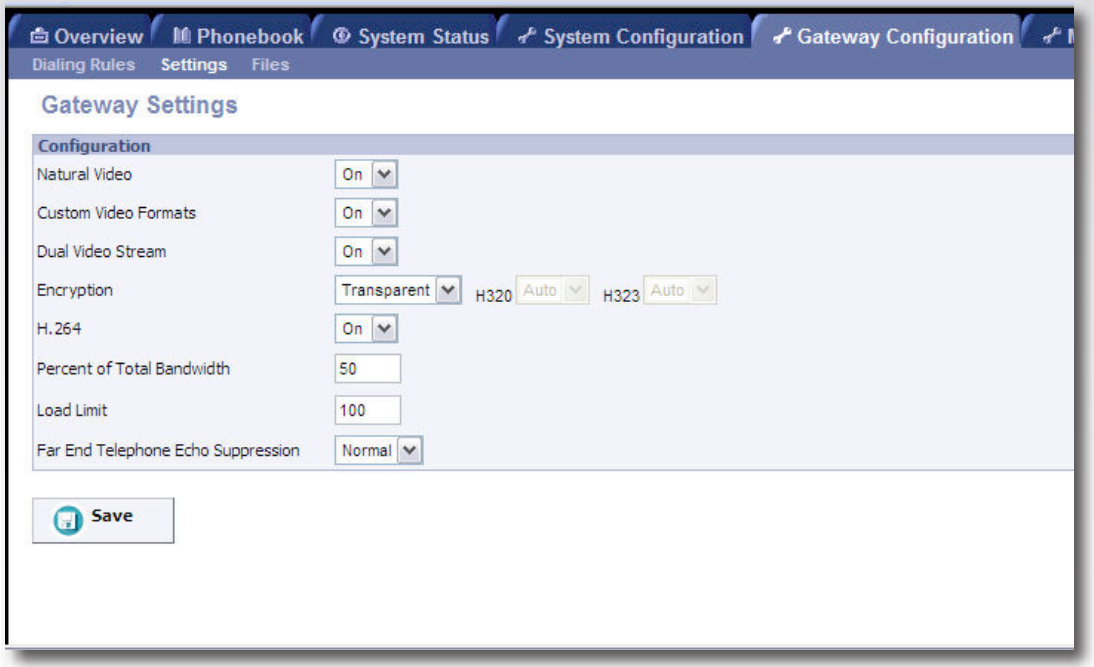
On: Encryption is required. Gateway connection is only established if encryption can be established on this side.

Off: No encryption. Gateway connection will be established without any encryption on this side.

Auto: Use encryption if available. Gateway will try to establish an encrypted connection, the connection will be established regardsless if encryption on this side is available.

External Crypto Device

A typical application of this feature is when an external crypto device is used on the ISDN interface. In this case an administrator may want to set the Gateway's encryption on the H.320 side to Off and on the H.323 side to On.



Save - Press the Save button to save changes.

Gateway Configuration, *cont...*

Gateway Settings

Interoperability

Experience has shown that some features - even though they are standardized - might cause interoperability problems with legacy video conferencing products.

By disabling features on this page, the Gateway can be used as a filter to ensure interoperability with legacy products.

H.264

On: Set to On to allow H.264 video capabilities to be transmitted through the Gateway.

Off: When set to Off no H.264 video capabilities will be transmitted.

Percent of Total Bandwidth

This setting defines the percentage of H.320 bandwidth that is dedicated to be used by the Gateway.

The default value is 0%

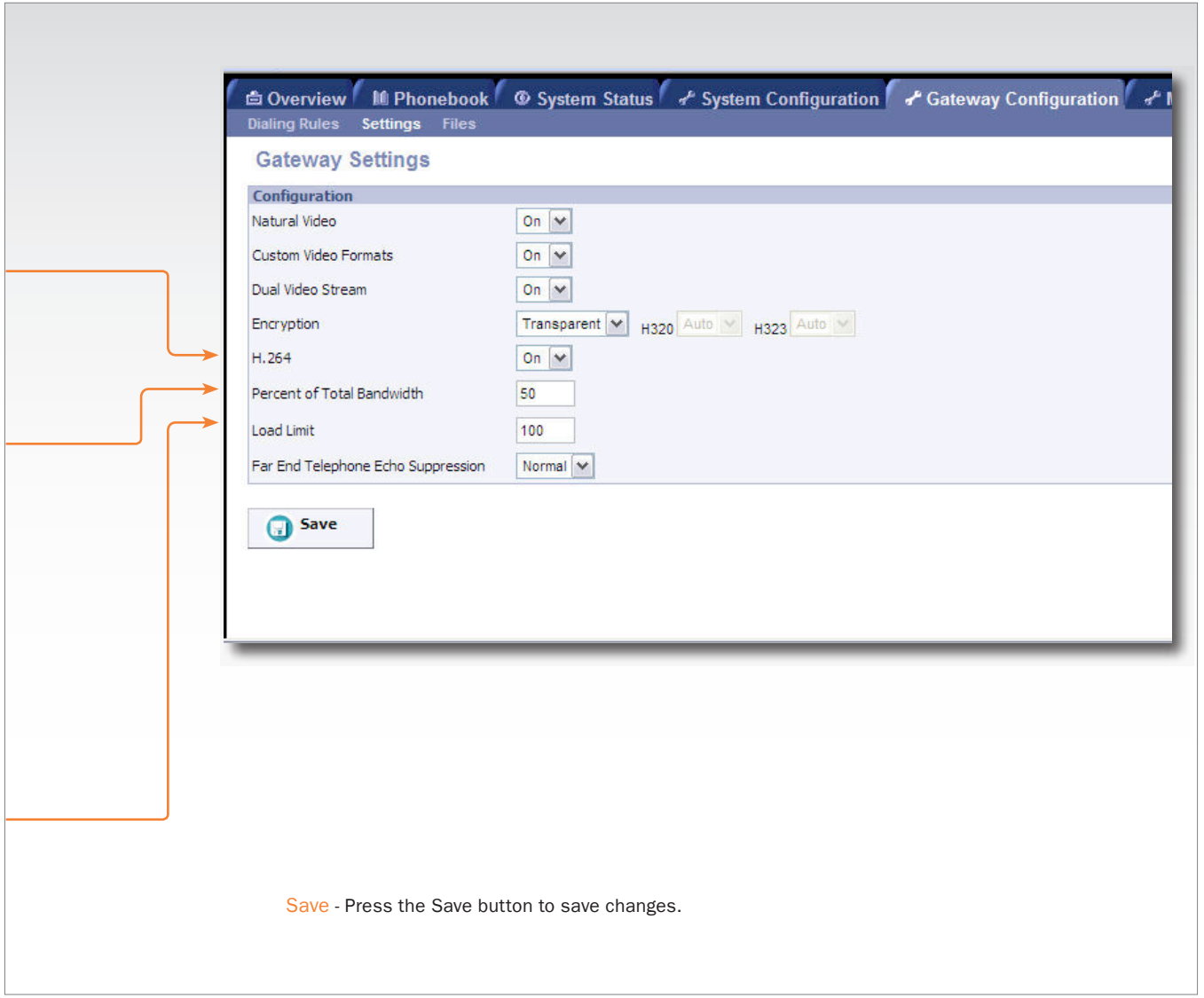
- If this value is set to 0 (zero) %, the Gateway will not have any H.323 resources available (same as actually disabling the Gateway).
- If this value is set to 100 (hundred) %, no Gateway calls will be refused as long as the MPS has resources available.
- However, using the value 100% will make it possible to use all bandwidth resources for Gateway calls not leaving any resources for MCU calls.

Load Limit

When the Resource Usage reaches the Load Limit, the Gateway will signal this state to the Gatekeeper. The Gatekeeper can then try to route IP/H.323 calls to other Gateways.

When using multiple Gateways this will maintain availability for incoming ISDN calls.

The value is set from 0 % to 100 %. Default value is 100 %



Save - Press the Save button to save changes.

Gateway Configuration, *cont...*

Gateway Settings

Interoperability

Experience has shown that some features - even though they are standardized - might cause interoperability problems with legacy video conferencing products.

By disabling features on this page, the Gateway can be used as a filter to ensure interoperability with legacy products.

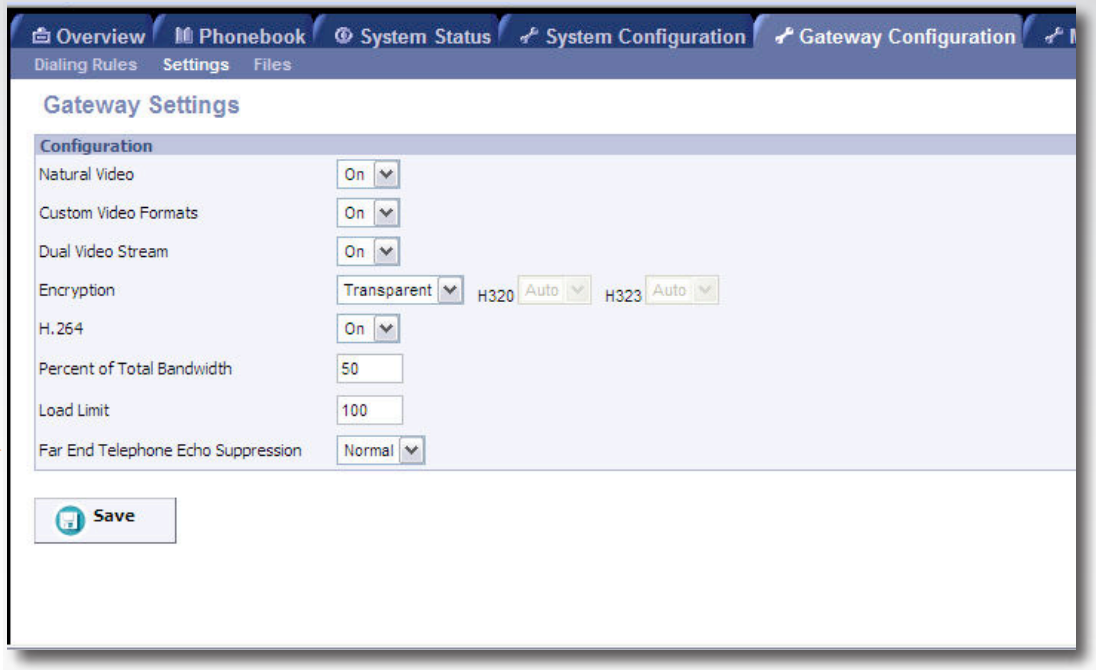
Far End Telephone Echo Surpression

Analogue telephone lines, speaker phones and telephone headsets may all cause echoes. The Far End Telephone Echo Suppression function eliminates some – or all – of the experienced echo.

Off: Set to Off to disable Far End Telephone Echo Suppression.

Normal: Weak echoes are removed with the Normal setting enabled.

High: Strong echoes are removed with the High setting enabled.



Save - Press the Save button to save changes.

Chapter 7

MCU configuration

The MCU Configuration section contains information about how to configure and use the MCU on the TANDBERG MPS. Reading this section makes you familiarize yourself with the configuration menu and functions important for the correct configuration and functioning of the MCU. You will also find information on how to configure dial in numbers for different types of conferences and about conference templates and file management.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

In this chapter...

MCU Configuration

The Dial-In Numbers menu overview

In the Dial In Numbers page you can configure numbers and/or aliases to create a conference and to dial into an already created conference.

There are different ways of configuring and setting up a conference depending on conference type and use.

Conferences can be created and started from the web interface or by dialling into the MPS, with or without a create password and access password.

The participants may be put in a Waiting Room for a predefined period until the conference starts and may have a limited number of login attempts for accessing the conference.

The TANDBERG MPS 800 (200) supports up to 40 (10) simultaneous conferences.

Single Dial In Number

The Single Dial In Number enables users to set up multiple conferences in an Ad Hoc manner using one single number to the MPS. By calling one single number into the MPS the user may:

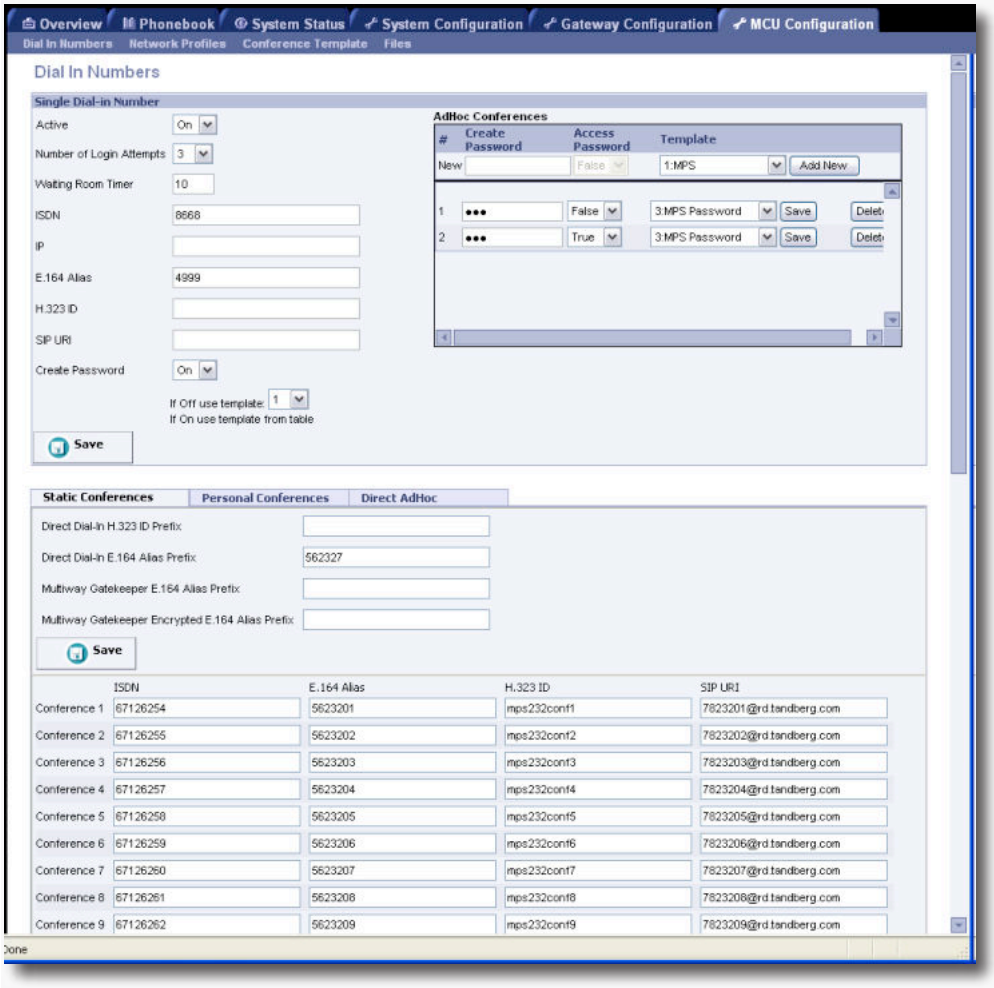
- Create a new conference on the fly
- Access an already created conference
- Be put on hold until a conference is created

- all by following the instructions on-screen on the videoconference system and by using the touch tones (DTMF) on the remote control.

Each conference may have a Create password and Access Password associated to the conference.

Note that Static, Personal and Direct Ad Hoc conferences may be accessed, but not created through the Single Dial In Number.

See Single Dial In Number for Ad Hoc Conferences for details.



Static Conferences

The Static Conferences configuration allows the users to define numbers and aliases for each conference on the MPS.

The conference is created from the web interface of the MPS.

See Static Conferences for details.

Personal Conferences

The Personal Conferences enables users to configure personal conferences with a fixed unique number and/or alias to dial into.

The creation of each conference can be password protected and a name can be associated to the conference.

See Personal Conferences for details.

Direct Ad Hoc Conferences

The Direct Ad Hoc field allows the user to configure their own conference based on a predefined prefix.

The conference is created by dialling a prefix followed by a user defined number and the conference participants call the same number to access the conference.

See Direct Ad Hoc Conferences for details.

MCU Configuration, *cont...*

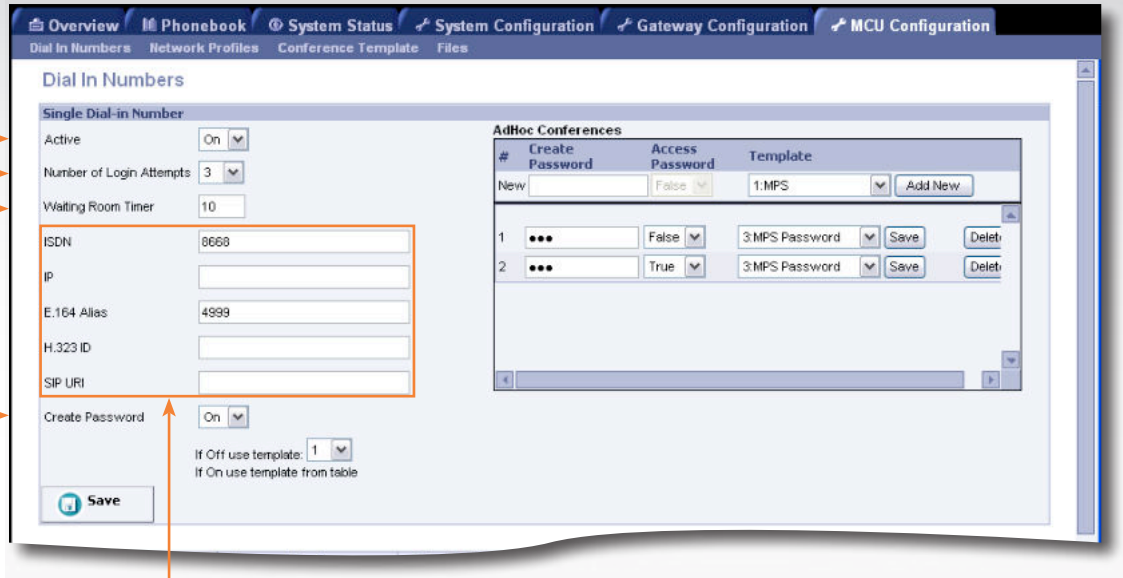
The Single Dial-In Number menu

The Single Dial In Number allows the MPS to set up multiple conferences, in an Ad Hoc manner. This is triggered by a user dialling in to a single number. The MPS will then provide a set of services that enables a user to:

- Create a new conference on the fly
- Access an already created conference
- Put on hold until a conference is created

In the Ad Hoc Conferences table Create Passwords, Access passwords and desired templates may be specified for a each conference.

Note that Static, Personal and Direct Ad Hoc conferences may be accessed, but not created through the Single Dial In Number.



Active

To make the configurations for Single Dial In Number functionality to take effect the Active setting must be set to On.

On: Set to On to enable the Single Dial In Number functionality.

Off: Set to Off to disable Single Dial In Number functionality.

Number of Login Attempts

Limits the number of login attempts a user may have to access a conference. Enter a number from 1 to 10.

Waiting Room Timer

When users dials into a conference which is not started the users are placed in a virtual waiting room (put on hold). The Waiting Room Timer limits the time (in minutes) a user may be placed in the waiting room.

Enter a value between 1 and 60. The default value is 10 minutes.

Enter the numbers to be used:

ISDN: The ISDN Single Dial In Number must be a valid PRI number of the TANDBERG MPS at hand.

IP: The IP address for Single Dial In Number must be a valid IP address on the TANDBERG MPS System Controller Card.

E.164 Alias: The E.164 Alias is a numeric H.323 E.164 Single Dial In Number. The TANDBERG MPS must be registered to a Gatekeeper.

H.323 ID: The H.323 ID is a alphanumeric H.323 ID Single Dial In Number. The TANDBERG MPS must be registered to a Gatekeeper.

SIP URI: The SIP URI is a alphanumeric SIP URI Single Dial In Number. The TANDBERG MPS must be connected to a SIP proxy.

Create Password

On: If set to On, configure the Ad Hoc Conferences to select templates and set passwords for each conference.

Off: If set to Off, select the Conference Template to be used from the drop down list.

MCU Configuration, *cont...*

The AdHoc Conferences menu

The Single Dial In Number allows the MPS to set up multiple conferences, in an Ad Hoc manner. This is triggered by a user dialling in to a single number. The MPS will then provide a set of IVR services that enables a user to:

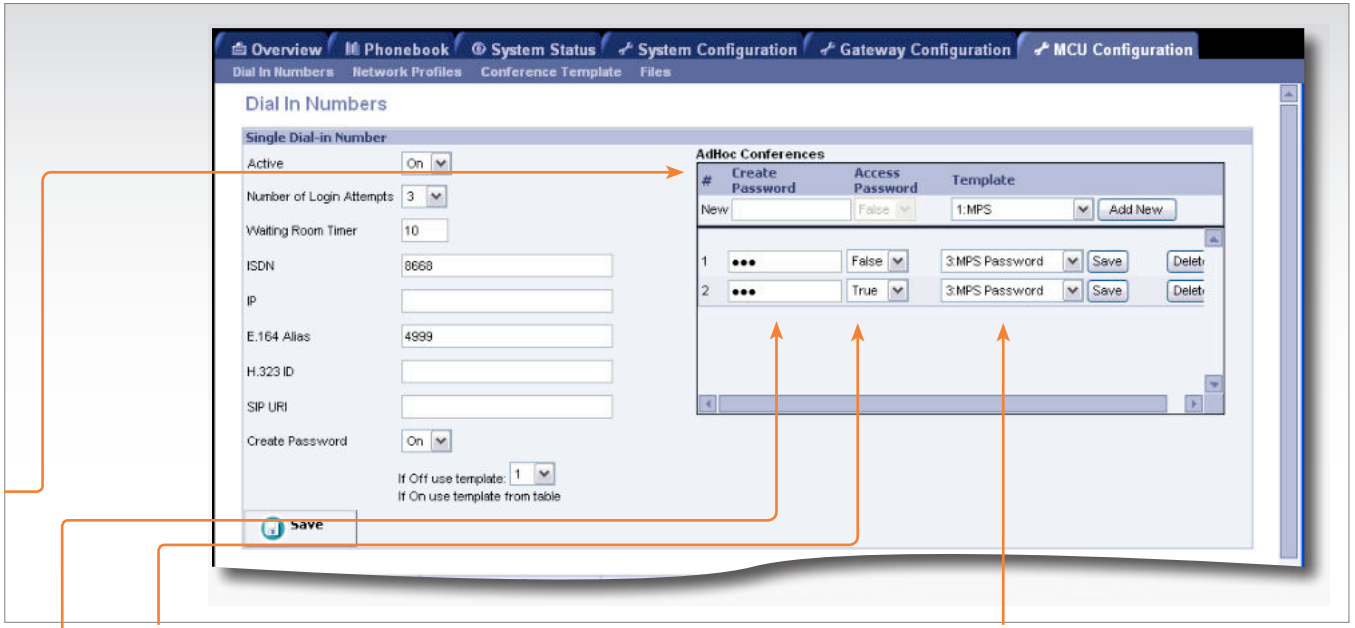
- Create a new conference on the fly
- Access an already created conference
- Put on hold until a conference is created

In the Ad Hoc Conferences table Create Passwords, Access passwords and desired templates may be specified for a each conference.

Note that Static, Personal and Direct Ad Hoc conferences may be accessed, but not created through the Single Dial In Number.

Ad Hoc Conferences

The Ad Hoc Conferences works in conjunction with the configuration of Single Dial In Number and may be used when the Create Password setting is set to On.



Create Password

The Create Password is used to authenticate the creator of the conference.

When you define more than one Create Password, please make sure the Create Password is unique.

This is a static password used to authenticate the owner of the the conference when the conference is created, and hence should not be distributed to anyone else than the conference owner.

If a Create Password is set, the first participant must be authenticated as the owner when dialling into the conference. If the field is left blank, there will be no authentication on the conference. The password must consist of numbers only as it is entered by using DTMF on the remote control when creating/accessing the conference.

Access Password

The Access Password is used to authenticate the particpants in a conference and is defined by the creator of the conference.

This field is only enabled when you have defined a Create Password.

True: If the Access Password is set to True, the creator is asked to provide a conference password, valid only for the conference at hand. Other participants in this conference must be informed on this password, to be able to access the conference.

False: If Access Password is set to False, the conference will not have any authentication.

Template

Up to 20 conferences may be defined in the conference list, one with Access Password and one without Access Password, for each of the 10 templates. You can also select the same template severeal times.

Select which template you would like the conference to use.

Add New: Press the Add New button to add a new item to the conference list.

Save

Press the Save button to save changes to your configuration.

Delete

Select a configuration from the list and press the Delete button to remove the selected configuration from the conference list.

MCU Configuration, *cont...*

The Static Conferences menu

The Static Conferences configuration allows the users to define prefixes, ISDN numbers, E.164 Aliases, H.323 ID's and SIP URI's for each conference on the MPS.

The number of conferences in the Static Conferences list correspond to the number of conferences in the Overview > MCU panel.

How to get access to a Static Conference

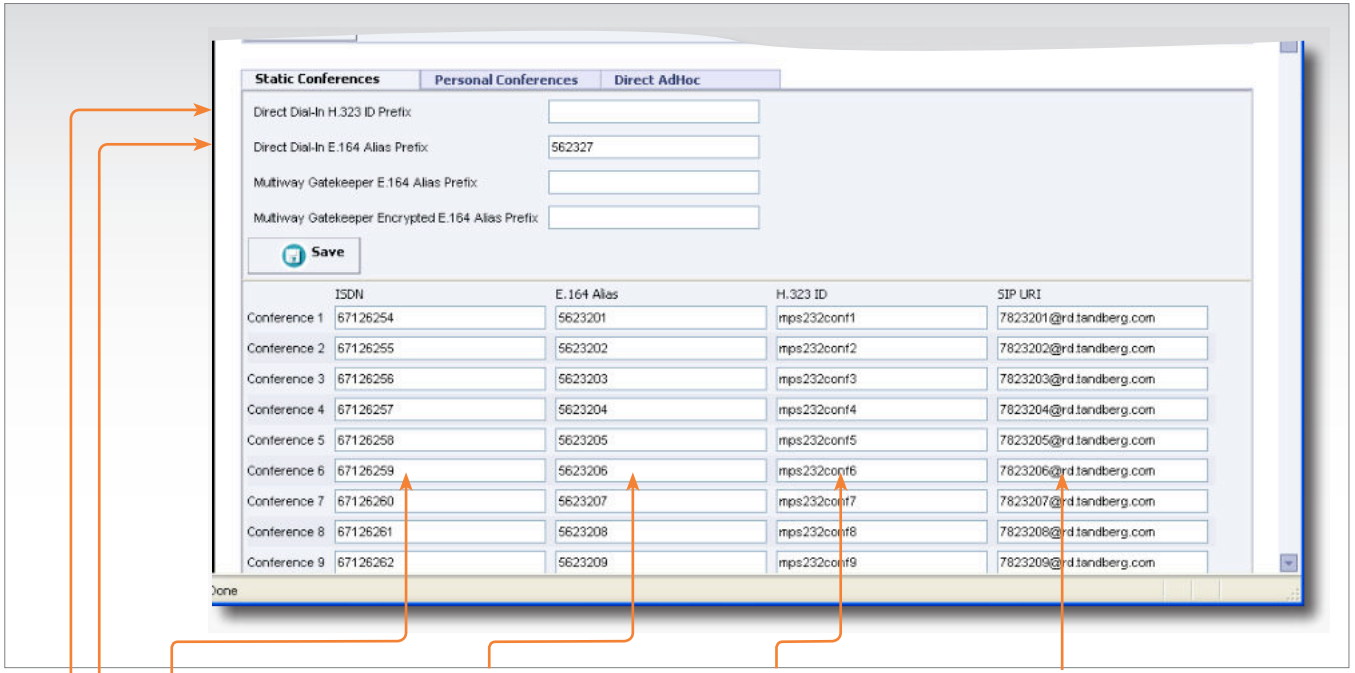
The conference must be created and started from the web interface of the MPS.

The participants can access a Static Conference by dialling a single number using the definitions in the Static Conferences, like ISDN, E.164 Alias, H.323 ID and SIP URI as the conference ID.

Waiting Room

When users dials into a conference which is not started the users are placed in a virtual waiting room (put on hold) for a predefined period of time until the conference starts.

The predefined period of time is defined in the Waiting Room Timer setting found under the Single Dial In Number configuration.



DID H.323 ID Prefix

The prefix value to be used together with H.323 ID numbers. The prefix will be registered on the connected Gatekeeper.

(DID - Dial In Direct)

DID E.164 Alias Prefix

The prefix value to be used together with E.164 Alias numbers. The prefix will be registered on the connected Gatekeeper.

Note that the DID H.323 ID Prefix and DID E.164 Alias Prefix are only used for adding extra dial in numbers for each conference.

E.164 Aliases

The H.323 E.164 numeric alias for each conference.

The TANDBERG MPS must be registered to a Gatekeeper.

ISDN

The ISDN number must be a valid PRI number of the TANDBERG MPS at hand.

H.323 ID

The H.323 alphanumeric ID for each conference.

The TANDBERG MPS must be registered to a Gatekeeper.

SIP URI

The SIP alphanumeric URI for each conference.

The TANDBERG MPS must be connected to a SIP proxy.

MCU Configuration, *cont...*

The Static Conferences menu

The Static Conferences configuration allows the users to define prefixes, ISDN numbers, E.164 Aliases, H.323 ID's and SIP URI's for each conference on the MPS.

The number of conferences in the Static Conferences list correspond to the number of conferences in the Overview > MCU panel.

How to get access to a Static Conference

The conference must be created and started from the web interface of the MPS.

The participants can access a Static Conference by dialling a single number using the definitions in the Static Conferences, like ISDN, E.164 Alias, H.323 ID and SIP URI as the conference ID.

Waiting Room

When users dials into a conference which is not started the users are placed in a virtual waiting room (put on hold) for a predefined period of time until the conference starts.

The predefined period of time is defined in the Waiting Room Timer setting found under the Single Dial In Number configuration.

MultiWay Gatekeeper E.164 Alias Prefix (removed from the web interface)

The H.323 E.164 numeric alias prefix for non-encrypted MultiWay (beta) is removed from the web interface. Works with F7.x and TANDBERG GW MultiWay (beta) only.

The setting is still available from the API using the command:

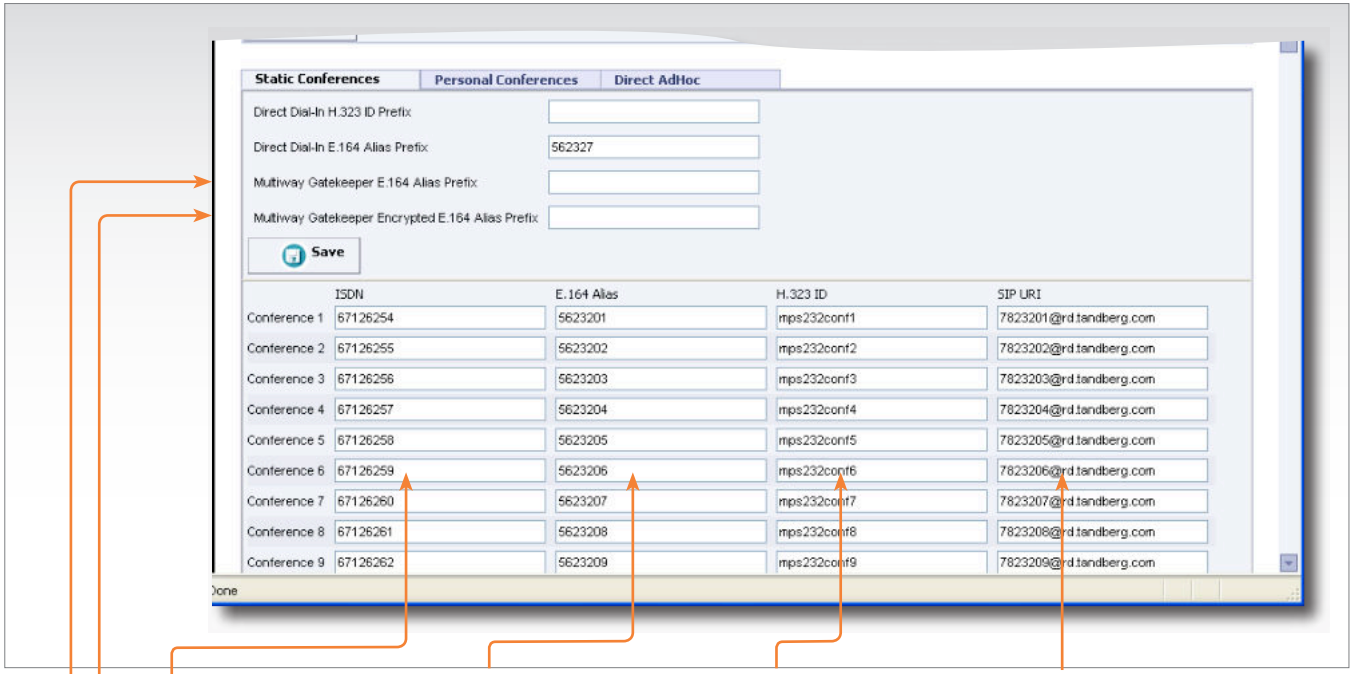
```
xConfiguration PrefixDialIn Gatekeeper
H323Alias
```

MultiWay Gatekeeper Encrypted E.164 Alias Prefix (removed from the web interface)

The H.323 E.164 numeric alias prefix for encrypted MultiWay (beta) is removed from the web interface. Works with F7.x and TANDBERG GW MultiWay (beta) only.

The setting is still available from the API using the command:

```
xConfiguration PrefixDialIn Gatekeeper
H323EncryptedAlias
```



E.164 Aliases

The H.323 E.164 numeric alias for each conference.

The TANDBERG MPS must be registered to a Gatekeeper.

ISDN

The ISDN number must be a valid PRI number of the TANDBERG MPS at hand.

H.323 ID

The H.323 alphanumeric ID for each conference.

The TANDBERG MPS must be registered to a Gatekeeper.

SIP URI

The SIP alphanumeric URI for each conference.

The TANDBERG MPS must be connected to a SIP proxy.

MCU Configuration, *cont...*

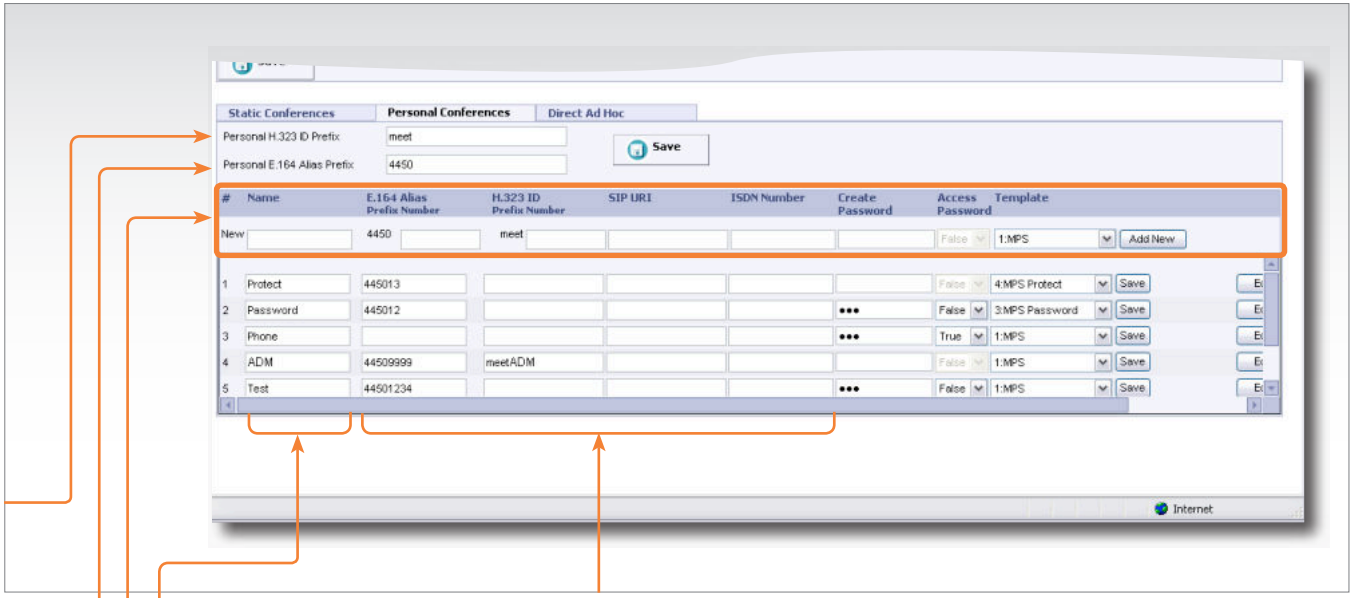
The Personal Conferences menu

The Personal Conferences allows you to specify one or more conferences with:

- Unique name
- Unique numbers and/or aliases
- Passwords
- Conference Template

When the personal conference is started, the participants defined in the Edit Participants configuration setup will automatically be called (excluding the starting participant himself if the incoming number is the same as the specified participant number).

This will ensure that the owner of the conference, or one or more other participants, always are called when someone is starting a personal conference.



Personal H.323 ID Prefix Number

Enter the prefix value to be used together with Personal conferences with H.323 ID numbers.

The prefix is registered on the connected Gatekeeper and must be entered before H.323 ID.

Personal E.164 Alias Prefix Number

Enter the prefix value to be used together with Personal conferences with H.323 E.164 Alias numbers.

The prefix is registered on the connected Gatekeeper and must be entered before E.164 Alias.

Add New Personal Conference

Fill in the required fields and press the Add New button to add a new Personal Conference configuration.

Name

Enter a Name to identify the the Personal Conference. This could be any text string.

Multiple Numbers

In the conference configuration you can define one or more numbers (H.320, E.164, H.323 or SIP) which will be used for dial in to a conference.

E.164 Alias

In this field, a unique numeric E.164 alias should be entered. Note that it is the corresponding Personal E.164 Alias Prefix that is registered to the Gatekeeper.

SIP URI

In this field, a unique SIP URI for the conference could be entered. You must be able to configure the SIP proxy to forward certain SIP URI to the MPS. Note that the full SIP URI should be entered and that functionality may be limited depending on the SIP server.

H.323 ID

In this field, a unique alphanumeric H.323 ID should be entered. Note that it is the corresponding Personal H.323 ID Prefix that is registered to the Gatekeeper.

ISDN Number

To be able to use ISDN numbers, make sure you have a PRI interface Card, available ISDN options and use only free numbers in your assigned PRI range.

MCU Configuration, *cont...*

The Personal Conferences menu

The Personal Conferences allows you to specify one or more conferences with:

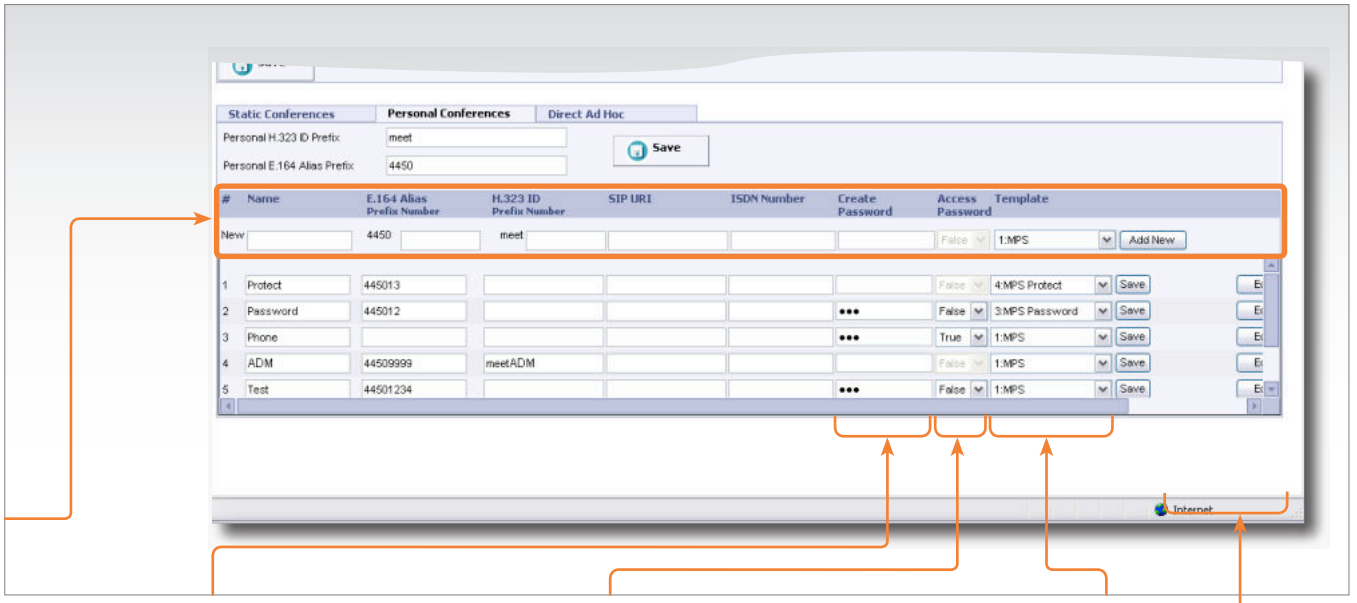
- Unique name
- Unique numbers and/or aliases
- Passwords
- Conference Template

When the personal conference is started, the participants defined in the Edit Participants configuration setup will automatically be called (excluding the starting participant himself if the incoming number is the same as the specified participant number).

This will ensure that the owner of the conference, or one or more other participants, always are called when someone is starting a personal conference.

Add New Personal Conference

Fill in the required fields and press the Add New button to add a new Personal Conference configuration.



Password Protection

The creation of each conference can be password protected as well as accessing the conference.

When a participant is dialling into a personal conference, which requires a Create Password and that is currently not started, the participant will be asked to start the conference with the create password or wait until it starts. The participant will automatically be connected when the conference is started.

No Password Protection

If no Create Password has been defined, the conference will start immediately when a participant calls in.

Create Password

The Create Password is used to authenticate the creator of the conference.

This is a static password used to authenticate the owner of the the conference when the conference is created, and hence should not be distributed to anyone else than the conference owner.

If a Create Password is set, the first participant must be authenticated as the owner when dialling into the conference. If the field is left blank, there will be no authentication on the conference.

The password must consist of numbers only as it is entered by using DTMF on the remote control when creating/accessing the conference.

Access Password

The Access Password is used to authenticate the particpants in a conference and is defined by the creator of the conference.

This field is only enabled when you have defined a Create Password.

True: If the Access Password is set to True, the creator is asked to provide a conference password, valid only for the conference at hand. Other participants in this conference must be informed on this password, to be able to access the conference.

False: If Access Password is set to False, the conference will not have any authentication.

Template

Select which Conference Template to be used for the Personal Conference.

Edit Participants

Press the the Edit Participans button to open the Add Participants dialog to add or remove participants.

Save

Press the Save button to save changes to your configuration.

MCU Configuration, *cont...*

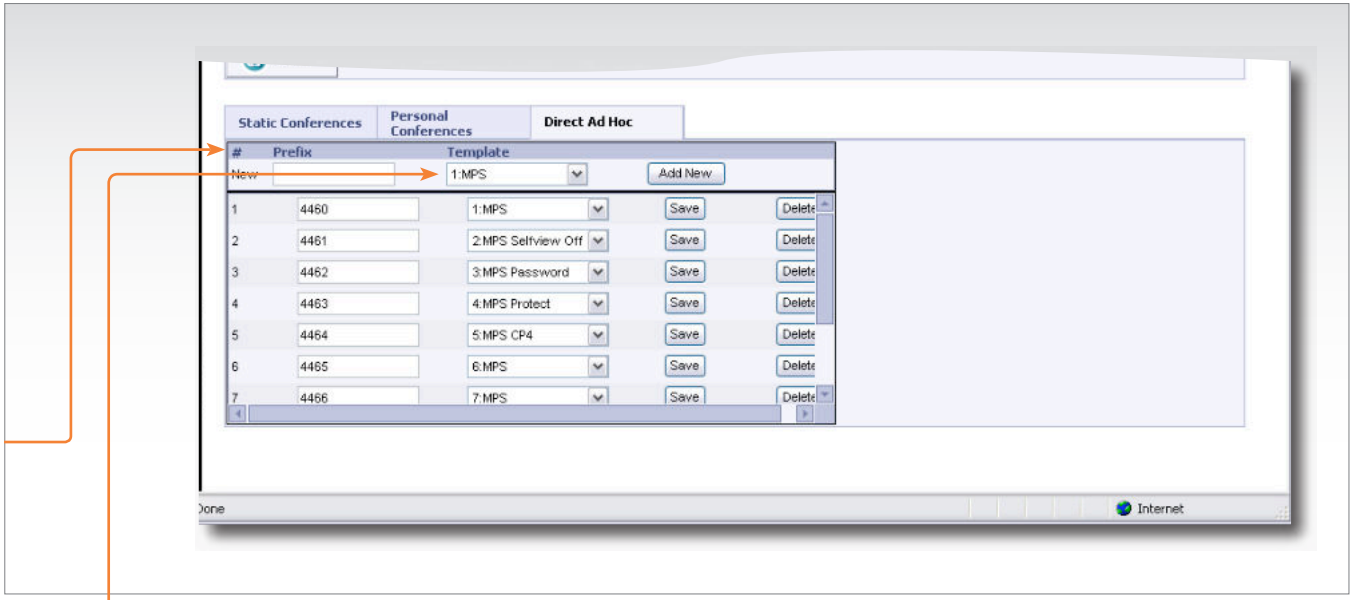
The Direct AdHoc Conferences menu

The Direct Ad Hoc conference is an alternative to Personal Conferences and the traditional Ad Hoc conferences.

With Direct Ad Hoc conferences you can define a prefix to be used together with several user defined numbers.

The user can easily create a conference by dialling: `<prefix + user defined number>`.

Up to 10 Direct Ad Hoc conferences may be defined. This allows one prefix for each of the 10 templates.



Prefix

Specify a Prefix to be used for this Direct Ad Hoc conference configuration.

A conference is created by dialling this prefix followed by a user defined number, typically the employee number for the owner of the conference.

Template

Select which Conference Template to be used for this Direct Ad Hoc conference configuration.

Example:

A company is divided into departments and groups:

- Each department and group has been given a Prefix.
- Each employee has a user defined number.

Start a Direct Ad Hoc conference:

1. A department has the prefix `<4450>`
2. The group manager has the user defined number `<21121>`
3. When the group is having their weekly status meeting the group manager dials `<445021121>` to create and start the conference.
4. To attend the meeting the other group members dials the same number.

Add New

Press the Add New button to add a Prefix and Conference Template for a new group of conferences to the configuration list.

Save

Press the Save button to save changes to your Direct Ad Hoc Conference configuration.

Delete

Press the Delete button to delete your Direct Ad Hoc conference configuration from the list.

MCU Configuration, *cont...*

The Network Profiles menu

Network Profiles

Network Profiles enables the user to specify a protocol and a service prefix for TANDBERG MPS dialling.

Name

Enter the name of the new Network Profile in the Name field.

Call Prefix

Enter the Call Prefix of the new Network Profile.

This could be 0 for external call (ref. example 1).

Network

Auto: IP addresses will select H.323. All other numbers will select ISDN.

H.320: The MCU will always use H.320 (ISDN) when this dial profile is selected.

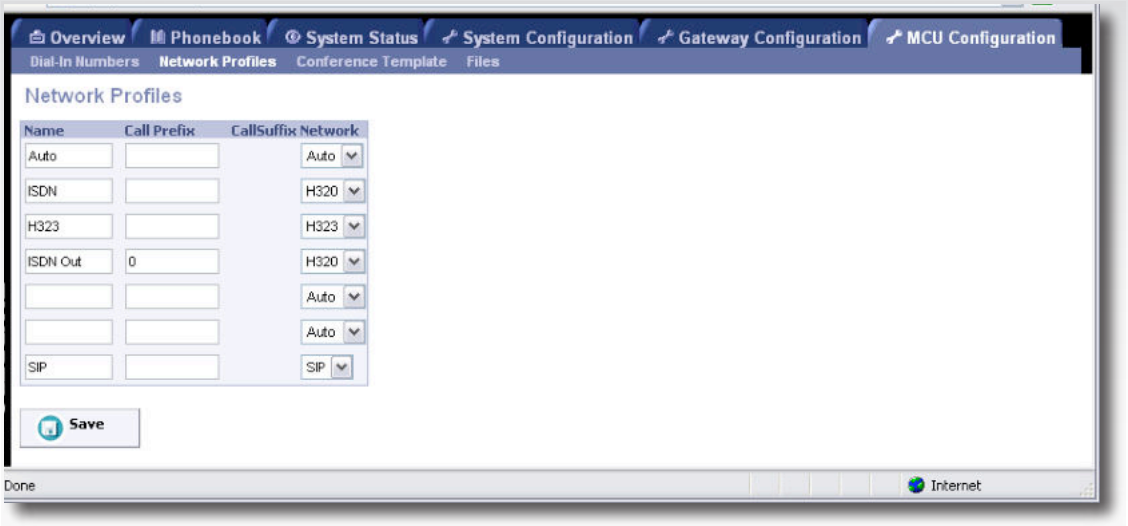
H.323: The MCU will always use H.323 (IP) when this dial profile is selected.

Example 1: Using a Call Prefix

Let us say you have to use the prefix 0 on ISDN in order to call outside your location.

Create a Network Profile:

1. Enter a Name; we will call it ISDN Out
2. Set Call Prefix to 0 and
3. Set Network to H.320 (ISDN).



Save:

Press the Save button to activate the new Network Profiles.

MCU Configuration, *cont...*

Conference Template Configuration

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Name

The conference name will be shown on the Conference Overview page and on the Conference Status page.

Maximum Conference Rate

Custom Selection: Specifiy the maximum possible call rate allowed in the conference. If a participant does not support this rate the MCU will connect at the highest rate possible.
Telephone: When Telephone is selected, an audio bridge will be created and no video participants will be able to join.

Default IP Net ID

Specifies which IP-network ID to use as default.

Restrict (56K)

Non-restricted and restricted calls are supported in the same conference. It is thus possible to select restrict for each call individually when dialling.

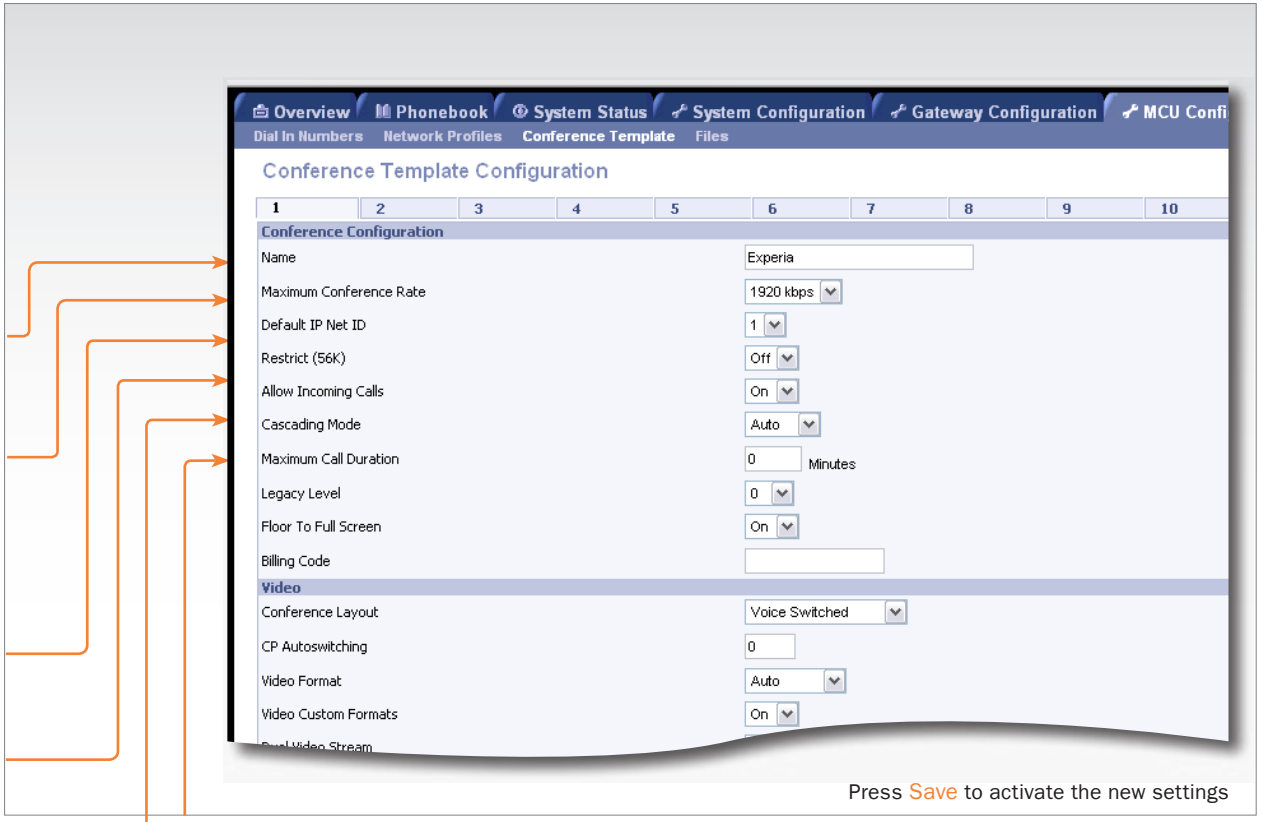
On: Set Restrict (56K) to On to make the MCU to set up restricted calls by default.

Off: Set Restrict (56K) to Off to make the MCU to set up non-restricted call and down-speed to 56 kbps if necessary.

Allow Incoming Calls

On: Set to On to allow incoming calls. Incoming call will be automatically answered.

Off: Set to Off to automatically reject all incoming calls.



Press Save to activate the new settings

Cascading Mode

Used to join two or more conferences together.

Auto: Set to Auto to automatically determine which conference is 'master' and which conference(s) are 'slave'. The 'master' conference will have control over the video layout. When left in 'Auto' mode, the conference dialling in to the other conferences, will become the 'master'.

Master: Set to Master when this conference is the one controlling the video layout for the whole conference. It is not recommended to have more than one 'master' in a conference.

Slave: Set to Slave when another conference manually has been assigned 'master'. The slave will be forced to Full Screen voice switched mode.

MCU Conference Template

Conference Template Configuration

Conference Template Configuration, 1–n

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Maximum Call Duration

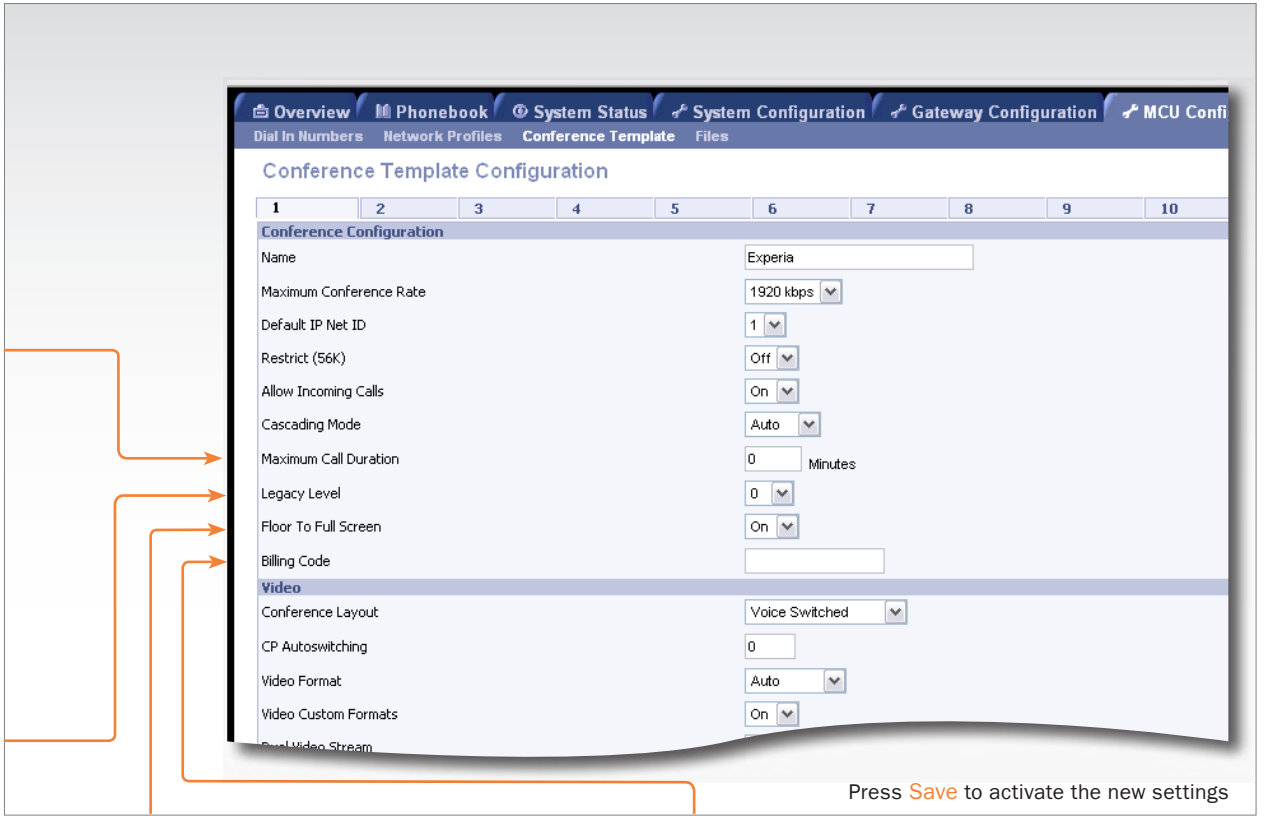
Determines the maximum duration (minutes) of the conference.

- All sites will be disconnected when the specified 'Max Call Duration' has been reached.
- 10 minutes, five minutes and one minute prior to this, a warning will be displayed to all the video participants in the conference, indicating the remaining time.
- The conference will remain active, after having timed out, allowing sites to dial in again and restart the conference timer.
- The conference administrator can extend the time.
- The timer for the max call duration will not begin until the first participant is connected.

Legacy Level

When connecting older videoconferencing endpoints to the MCU, problems can occur since older equipment sometimes do not handle modern capabilities.

- When set to 0-7: All capabilities are sent from the MCU
- When set to 8-14: The H.264 is disabled.
- When set to 15: The only capabilities sent for level 15 are H.261, G.711 and G.722.



Floor to Full Screen

On: When set to On, the participant requesting the floor will be shown in full screen to all the other video participants, regardless of current speaker. The same will happen if the conference administrator assign floor to a site.

Off: When set to Off, the participant requesting the floor will be shown in the largest sub-picture if there is one in the selected layout.

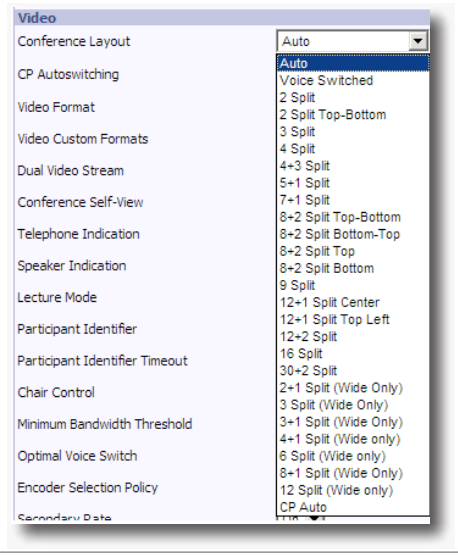
Billing Code

When defining a conference, a specific billing code can be assigned to the conference.

All calls in this conference will be associated with this billing code. This will allow management tools, such as the TANDBERG Management Suite, to use the code for billing purposes.

MCU Conference Template, *cont...*

Conference Layouts



Conference Layout

Auto: When set to Auto the most suitable conference layout will automatically be selected depending on the total number of participants in the actual conference.

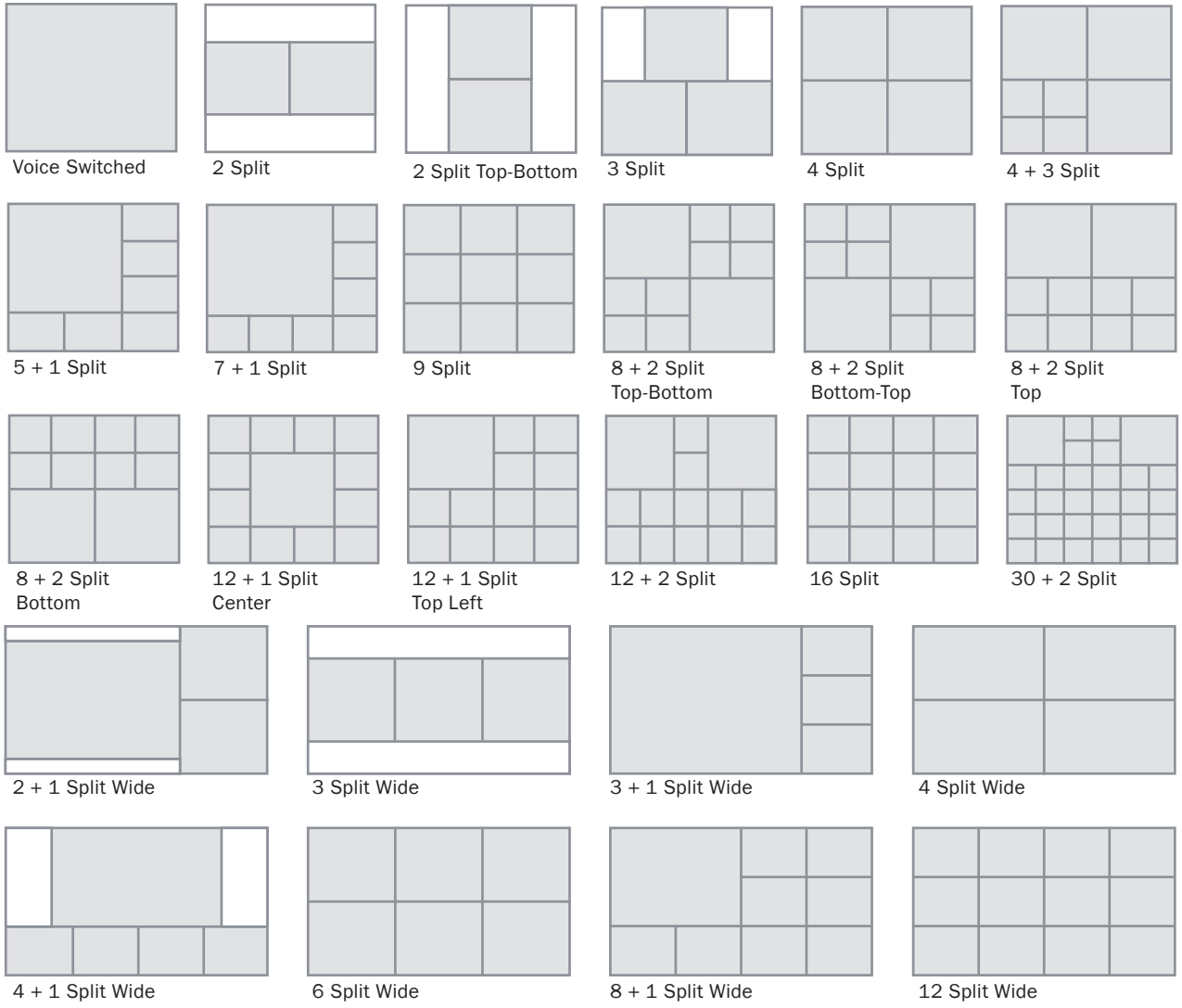
Voice Switched: Full Screen voice switched will show the current speaker in full screen to all the other participants, regardless of how many participants there are in the conference. Current speaker will see the previous speaker.

Custom Selection: Select a specific Conference Layout for the conference. The different selections are illustrated to the right.

CP Auto: When set to CP Auto there will be a dynamic change in layout dependent on the number of sites in the conferense. The CP Auto will start with VS->CP4->CP9->CP16.

Show Current Speaker

The screen will be split into a specified number of sub-pictures. The currently speaking participant will be shown in the largest sub-picture in asymmetric layouts. With fewer participants than the total number of sub-pictures, the empty sub-pictures will be black. If there are more participants than the total number of sub-pictures, only the last speakers will be displayed.



MCU Conference Template, *cont...*

Video settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

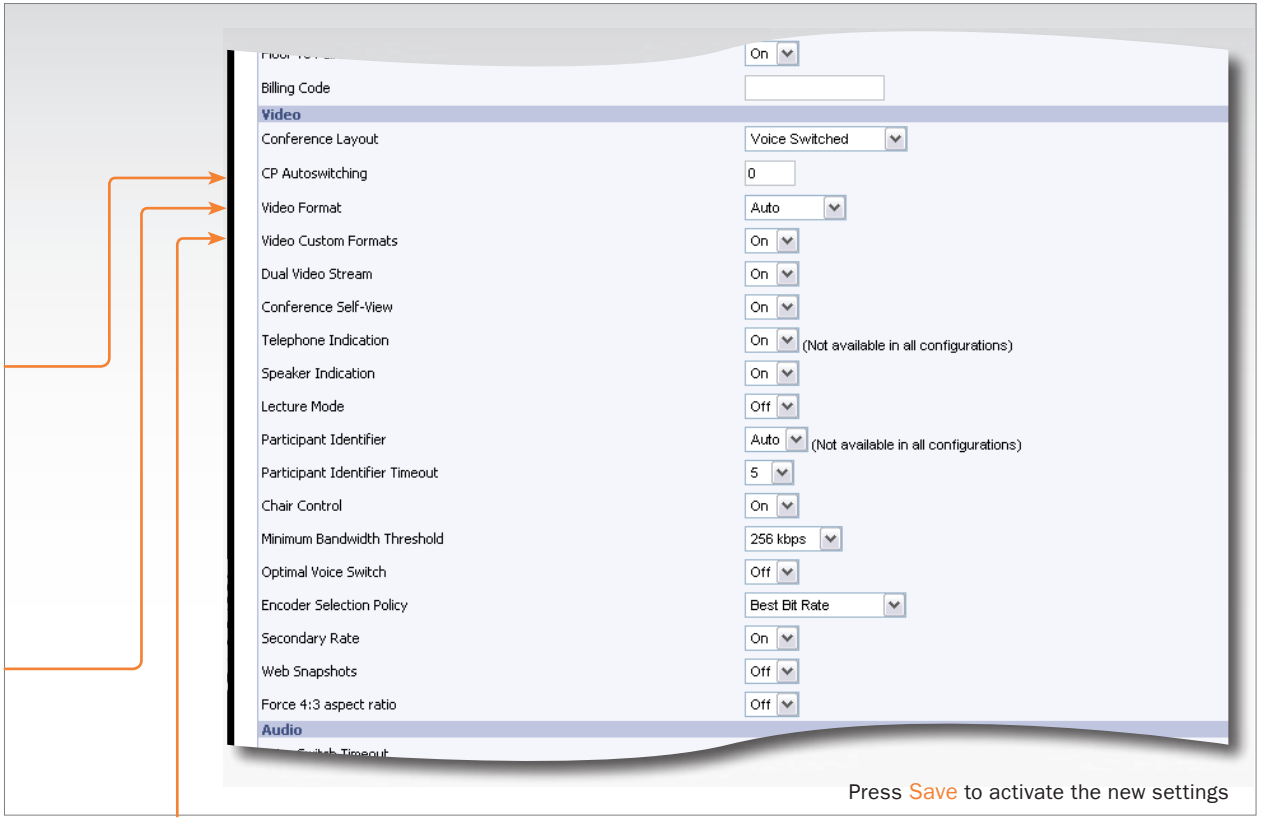
CP Autoswitching

The CP Autoswitching enables to swap non speaking sites with the least active sites in the picture. This lets you see all participants in a conference, even if they are not speaking.

CP Autoswitching can be set to a value between 0 seconds (default) and 60 seconds. The number of seconds denotes how long each of the remaining participants shall be displayed on the screen.

If set to 0 seconds the CP Autoswitching will be disabled.

Note that the CP Autoswitching will be performed in another picture if one or more of the participants speak.



Press Save to activate the new settings

Video Format

Defines the video format for Continuous Presence (CP) mode.

Auto (Best Impression[TF]) In Continuous Presence mode the MPS will select Motion (CIF) if the call rate is below 256 kbps and sending 4:3 aspect ratio. When sending 16:9 aspect ratio the MPS will select Motion (w288p) if the call rate is below 512 kbps. At call rates of 256 kbps and higher the MPS will select Sharpness (4CIF) when sending 4:3 aspect ratio. When sending 16:9 aspect ratio the MPS will select Sharpness (w576p) at call rates of 512 kbps and higher.

Motion: Set to Motion to prioritize motion and show up to 30 fps in CIF resolution and transmit the highest common format, preferably H.264 CIF when sending 4:3 aspect ratio or H.263+ w288p when sending 16:9 aspect ratio.

Sharpness: Set to Sharpness to prioritize crisp and clear picture and transmit the highest common format, preferably H.263+ 4CIF when sending 4:3 aspect ratio or H.263+ w576p when sending 16:9 aspect ratio.

In Full Screen Voice Switched Conference layout, the MCU will prioritize H.264 CIF as the highest common format.

Video Custom Formats

On: Set to On to support custom formats, such as SIF and VGA resolutions. It allows true resolution to be maintained, rather than being scaled to another format. This is of particular benefit to users of NTSC and VGA resolutions, ensuring that their images are not scaled to fit with the PAL standard.

Off: Set to Off when support for custom formats are not needed.

MCU Conference Template, *cont...*

Video settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Dual Video Stream

The MCU supports DuoVideo[TF], H.239 and BFCP.

On: Set to On to enable Dual Video Stream for this conference. Both DuoVideo[TF], H.239 are supported in the same conference.

Off: When set to Off, Dual Video Stream will not be supported in this conference.

Conference Selfview

On: Set to On to enable Conference Selfview. The user will see himself in the picture when more than one participant is in the conference.

Off: Set to Off to disable Conference Selfview.

Telephone Indication

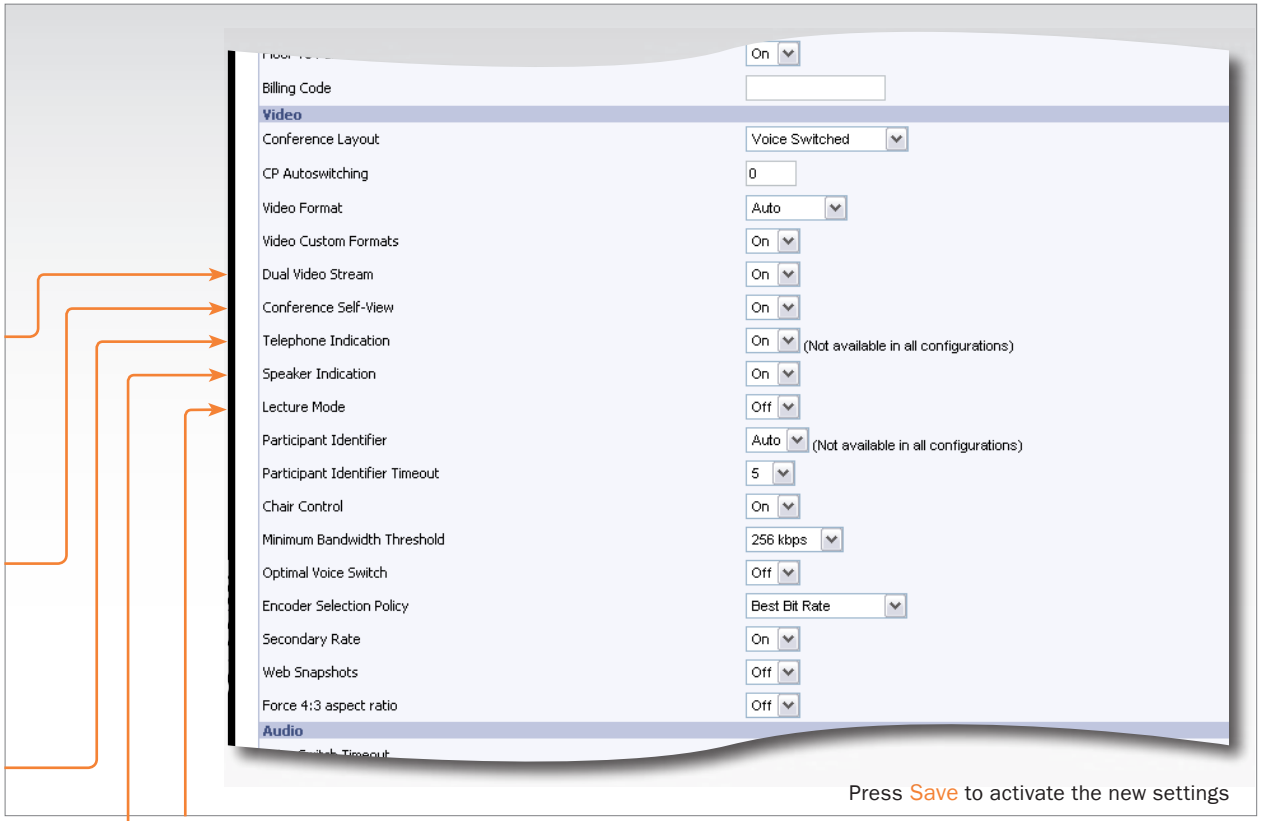
On: Set to On to enable a Telephone Indicator to be displayed when there are telephone (audio only) participants connected to the conference. When the telephone participant is speaking the indicator will be outlined.

Off: Set to Off to disable the Telephone Indicator to be displayed.

Speaker Indication

On: Set to On to enable a Speaker Indicator (a coloured line) to be displayed around the sub-picture that will indicate who is the currently speaking participant.

Off: Set to Off to disable the coloured line to be displayed.



Lecture Mode

On: Set to On to enable the Lecturer to be displayed in full screen to the other participants.

- The Lecturer is the participant which is assigned floor.
- The Lecturer will see a scan of all the participants in a full screen view or one of the supported sub-picture views.

Off: Set to Off to disable the Lecturer, the participant which is assigned floor, to be view in full screen.

MCU Conference Template, *cont...*

Video settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Participant Identifier

Auto: Set to Auto to let the System Name of a participant to be displayed the number of seconds set in Participant Identifier Timeout.

On: Set to On to enable the System Name for each participant to be displayed in the picture during the conference.

Off: Set to Off to disable the System Name to be displayed.

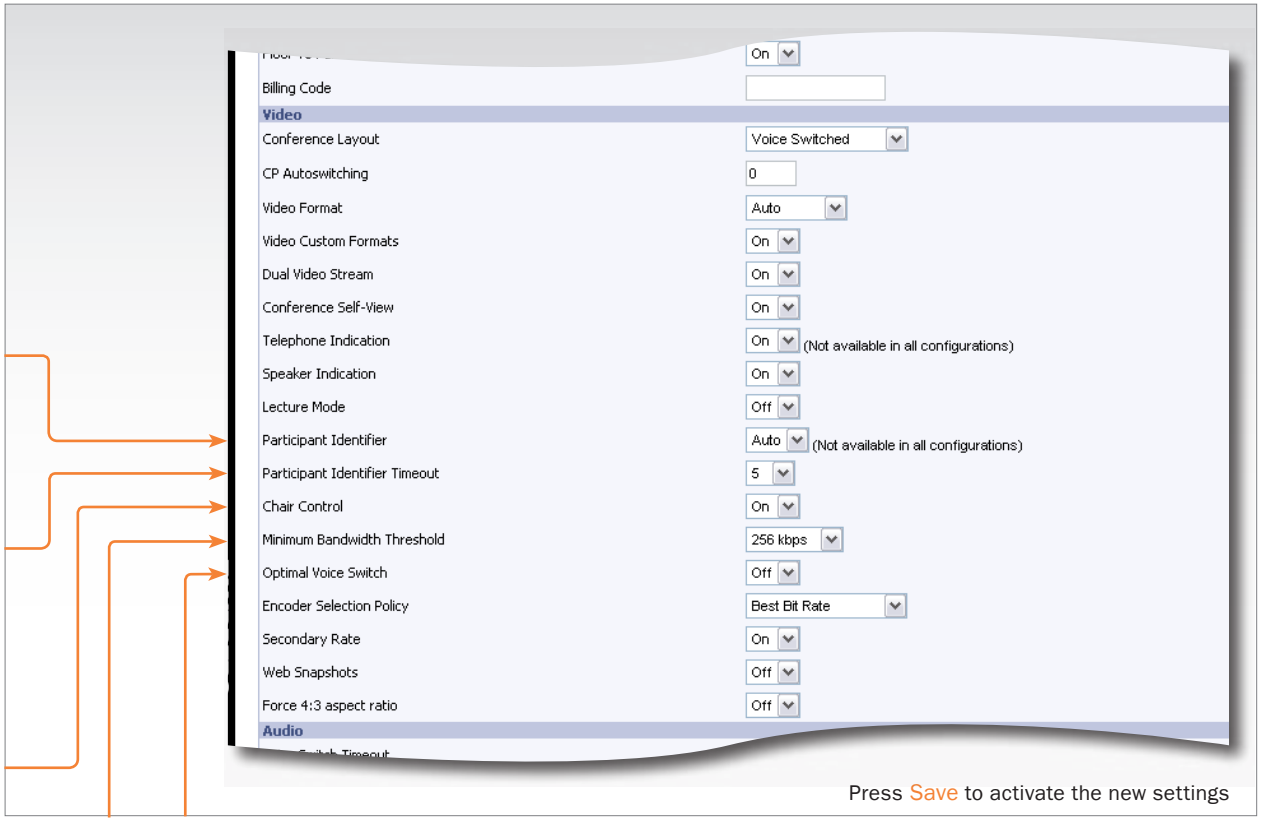
Participant Identifier Timeout

Set the number of seconds (1 - 30 seconds) the Participant Identifier will be visible, if set to auto. The identifier will re-appear at every picture changing event.

Chair Control

On: Set to On to enable Chair Control. The conference supports H.243 and BFCP Chair Control functionality initiated from the participants connected to the conference.

Off: Set to Off to disable Chair Control,



Minimum Bandwidth Treshold

If a participant calls in with a lower bandwidth than the Minimum Bandwidth Threshold, the participant will receive audio only (not live video) as well as a poster saying the bandwidth is to too low. After 10 seconds the participant will receive low rate video.

- The Minimum Bandwidth Threshold can be modified during a conference.
- The system will move calls below the defined Minimum Bandwidth Threshold to a low rate encoder.

NOTE: Once a participant is moved to the low rate encoder, they will not be moved back even if the Minimum Bandwidth Threshold is lowered.

Optimal Voice Switch

On: Set to On to enable Optimal video format in Voice Switch mode, if the connected endpoints allows this.

Off: When set to Off there will be normal transcoding when doing Voice switch.

1. Optimal Voice Switch is only avaliable on IP.
2. Icons and text are not avaliable when set to On.

MCU Conference Template, *cont...*

Video settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Encoder Selection Policy

Best Bit Rate: Set to Best Bit Rate to make the MPS prioritize the video quality for sites based on bit rate. The system will move participants with a Low Video Rate to a secondary encoder, if it is available. If no sites are moved, the system will move sites with Low Video Standard.

Best Video Standard: Set to Best Video Standard to make the MPS prioritize sites based on video standard. The system will move participants with a Low Video Standard to a secondary encoder, if it is available. If no sites are moved, the system will move sites with Low Video Rate.

Best Resolution: Set to Best Resolution to make the MPS prioritize the video quality for sites based on resolution. The system will move participants with a Low Resolution to a secondary encoder, if it is available. If no sites are moved, the system will move sites with low video rate.

Secondary Rate

On: Set to On to enable Secondary Rate. The conference will support two outgoing bandwidths if needed, in addition to the low rate video.

Off: Set to Off to disable Secondary Rate.

- Video settings continued...

Press the Save button to activate the new settings

Web Snapshots

The web snapshots are shown in the upper right corner of the web interface, and will show snapshots of the video from the participants and dual video stream. The snapshots are updated in accordance to the refresh rate (placed above the snapshot).

On: Set to On to enable Web Snapshots. When set to On the Conference Snapshot and Dual Video Stream Snapshot will show the video transmitted from the MCU to the participants.

Off: When set to Off a picture will appear telling that the Web Snapshots are disabled.

Force Aspect Ratio 4:3

This option modifies the default aspect ratio selection policy in a conference with a mix of 16:9 and 4:3 endpoints. The setting has no impact on switched video streams (see oVS).

On: When set to On, the video aspect ratio for the conference will be forced to 4x3. This differs from the default setting of the MPS, where the aspect ratio will be 16x9 as long as all participating end points can support 16x9. However, the default setting does not take into account situations where the Codec supports 16x9, but the screen is 4x3. A common example of such a scenario is for large rooms with LED's or other large screens.

Off: When set to Off (default) the aspect ratio will be based on the capabilities of the endpoints.

MCU Conference Template, *cont...*

Audio settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Voice Switch Timeout

Defines the number of seconds between 1 and 10, a participant must speak before it gets the speaker indication and is shown as the speaker to the other endpoints.

A long timeout might be suitable in noisy environments and in conferences with many participants.

Audio Leveling (AGC)

Ensures that all participants will receive the same audio level from all other participants, regardless of the levels transmitted. AGC - Automatic Gain Control.

In most conferences, the participants will speak at different levels. As a result, some of the participants are harder to hear than others. The Audio Leveling corrects this problem by automatically increasing the microphone levels when “quiet” or “distant” people speak, and by decreasing the microphone levels when “louder” people speak.

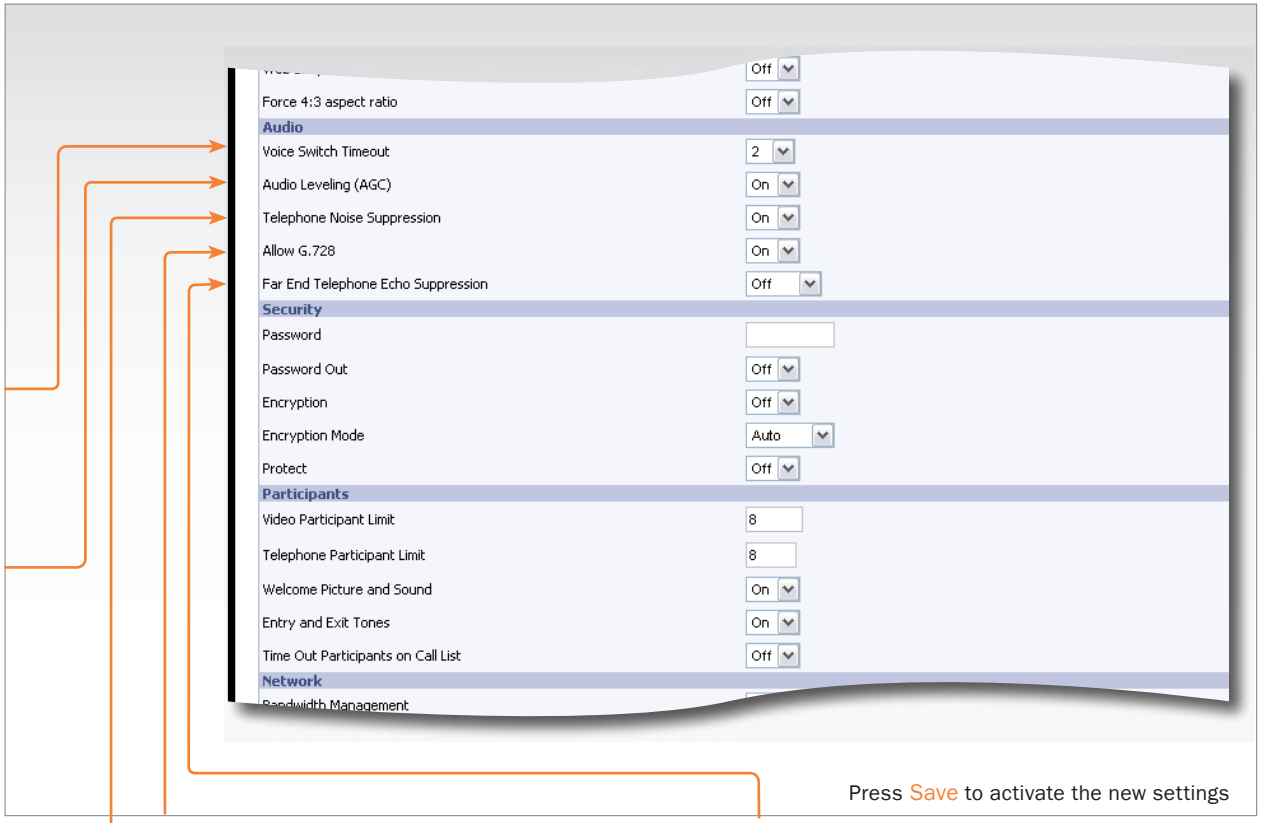
On: When set to On the MCU maintains the audio signal level at a fixed value by attenuating strong signals and amplifying weak signals. Very weak signals, i.e. noise alone, will not be amplified.

Off: Set to Off to disable Audio Leveling (AGC).

Telephone Noise Suppression

On: Set to On to enable Telephone Noise Suppression. Attenuates the noise which normally is introduced when adding mobile phones to a conference. The background noise normally heard when the telephone participant is not speaking will be attenuated.

Off: Set to Onffto disable Telephone Noise Suppression.



Allow G.728

On: The MCU supports high quality audio even on low call rate. On low call rate the MCU will prioritize G.722.1. The video participants which do not support G.722.1 will receive low quality audio G.728 instead when Allow G.728 is set to On.

Off: To ensure high quality audio on low call rate, set Allow G.728 to Off. Then video participants which are not able to support G.722.1, will receive G.722 instead.

Far End Telephone Echo Suppression

Analog telephone lines, speaker phones and telephone headsets may all cause echoes. The Far End Telephone Echo Suppression function eliminates some or all of the experienced echo.

Off: Set to Off to disable Far End Telephone Echo Suppression.

Normal: Set to Normal to remove weak echoes.

High: Set to High to remove strong echoes.

MCU Conference Template, *cont...*

Security settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Password

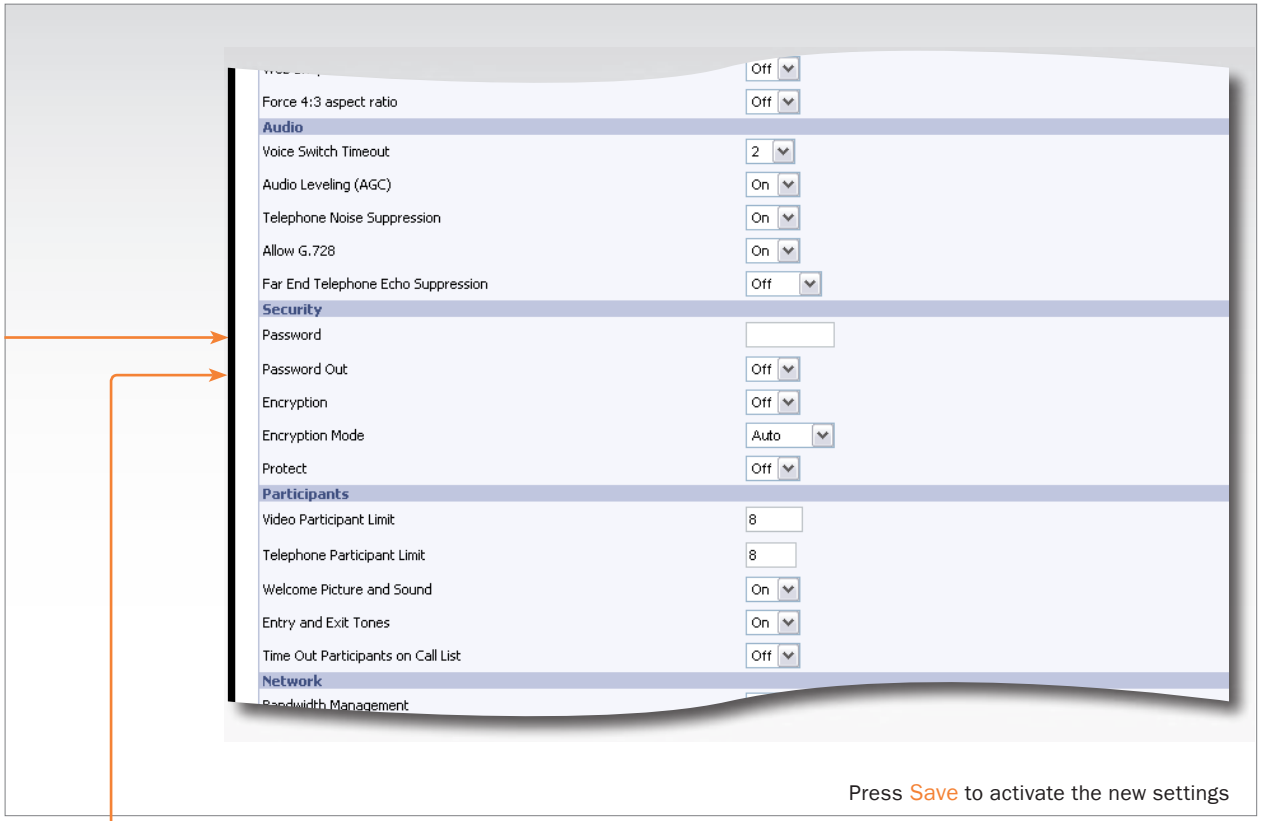
To ensure only authorized participants are able to join this conference you can set a password. Then the participants must enter the password to join this conference. The password can be numerical only.

- When dialling into a password protected conference, the participant is met with the 'Password Enquiry' screen and sound, asking the participant to enter a password. This can be performed via a menu generated by the videoconferencing system (H.243 Password) or via DTMF (telephone) tones.
- Until the correct password is entered, the participant will not be able to hear or see any of the other participants. After entering the correct password and confirming (typically by pressing 'OK' or the hash key), the participant will join the conference.
- Should the password be incorrect, the participant is met with the 'Password Incorrect' screen and after a few seconds, the 'Password Enquiry' screen and sound appear again. If the participant enters a wrong password three times, the participant will be disconnected.
- With no password entered in this field, participants can join the conference without entering a password

Password Out

On: When set to On and dialling out from a password protected conference, the participant is met with the 'Password Enquiry' screen and sound, asking the participant to enter a password. This setting can be used to ensure that only authorized participants are able to join the conference also when dialling out from the conference.

Off: When set to Off no password is required when dialling out.



MCU Conference Template, *cont...*

Security settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Encryption

On: When set to On all participants in the conference must support DES or AES encryption (available on all TANDBERG endpoints using software version B4.0 or later).
Participants not supporting encryption will be shown the 'Encryption Required' screen for 60 seconds before they are disconnected from the conference.

Off: When set to Off the conference will not be encrypted.

In a secure conference, there is no support for telephone participants.

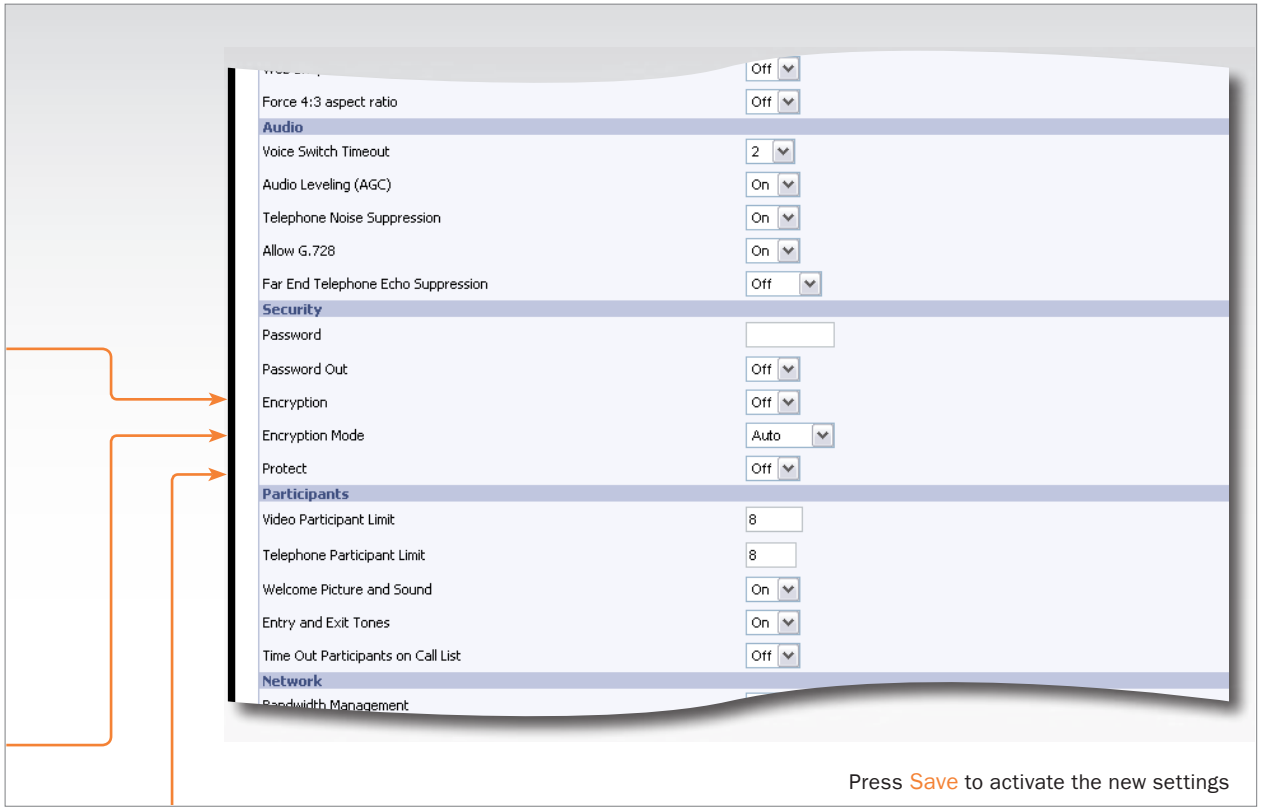
Encryption Mode

This settings only applies if Encryption is set to On (see above).

Auto: Set to Auto to use the highest level of encryption available on each of the participants connected in the conference. This means that there can be a mix of DES and AES encrypted connections in the same conference.

AES-128: Set to AES-128 to allow only participants with AES 128 bit encryption capabilities. Participants without this capability will not be able to join the conference.

DES: Set to DES to allow only participants with DES 56 bit encryption capabilities. Participants without this capability will not be able to join the conference.



Press Save to activate the new settings

Protect

On: When Protect mode is set to On:

1. Only predefined Protected Numbers are allowed to join this conference.
2. The Protected Numbers field will be shown, and Protected Numbers can be configured from the Dial In Configuration in the MCU Conference Overview page.

Off: Set to Off to disable the Protect mode.

For further information on Protected Numbers, see Protected Numbers in the Dial In Configuration in the Manage an Ongoing Conference section.

MCU Conference Template, *cont...*

Participants settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Video Participant Limit

Defines the maximum number of Video Participants allowed in the conference and reserves the number of needed Advanced Video Option ports for this conference.

Values: 0 - 40 for MPS 200 and 0 - 160 for MPS 800.

Telephone Participant Limit

Defines the maximum number of Telephone Participants allowed in the conference.

Values: 0 - 32 for MPS 200 and 0 - 48 for MPS 800.

Welcome Picture and Sound

On: When set to On a Welcome screen and audio message will be shown to each new participant of the conference.

Off: Set to Off to disable the Welcome screen and audio message.

Entry and Exit Tone

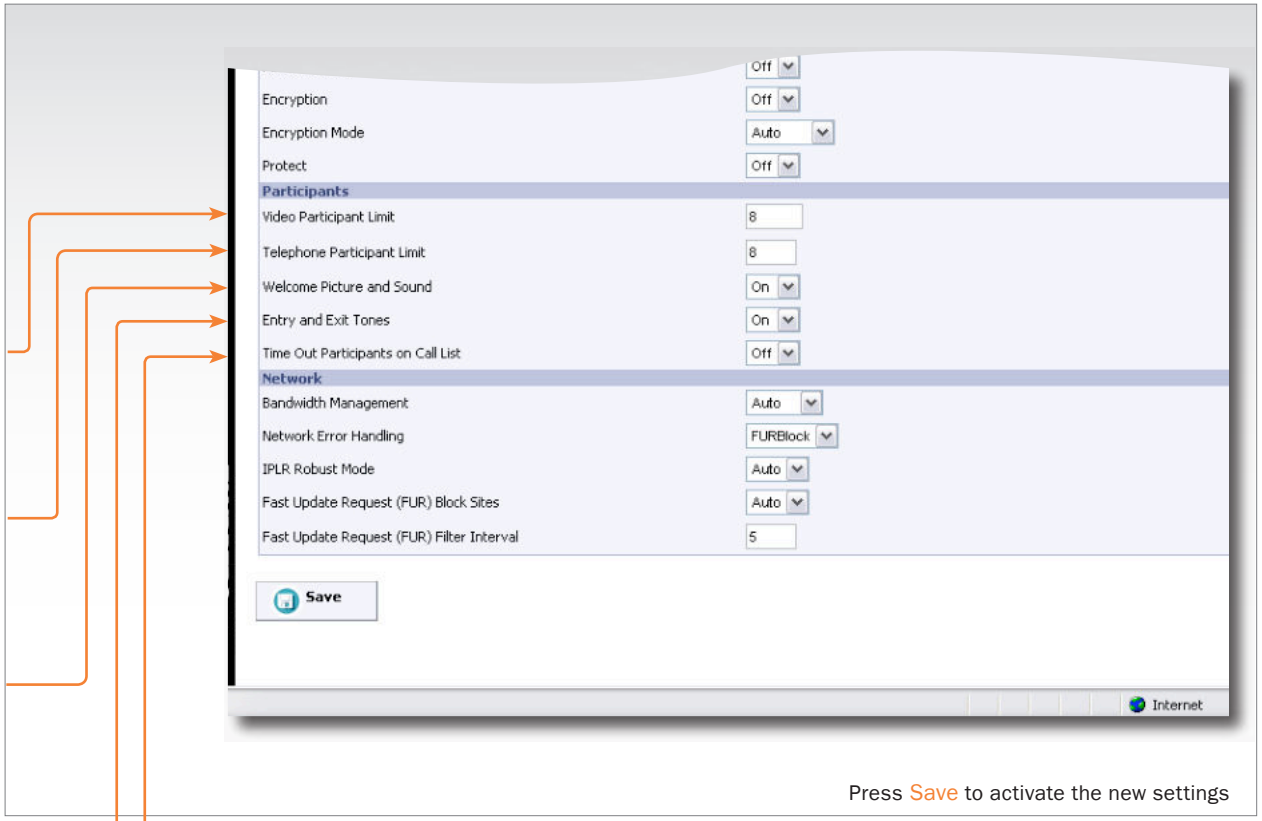
On: When set to On a tone signal will be heard each time a participant is entering or leaving the conference.

Off: Set to Off to disable the Entry and Exit Tone.

Timeout Participants from Call List

On: When set to On all participants that has been disconnected from the conference will be cleared from the Call List within 2 minutes.

Off: Set to Off to disable the Timeout Participants from Call List.



Press Save to activate the new settings

MCU Conference Template, *cont...*

Network settings

Conference Template Configuration

The Conference Templates are found in the MCU Configuration tab.

The predefined settings will be used as default settings when creating new conferences based on this specific template. TANDBERG MPS enables the setup of 10 different conference templates.

The conference template contains the same settings as described in the Overview > MCU > Create Conference section.

Bandwidth Management

Manual: Disables automatic regulations of sites to Low rate encoder, based on video rate reports.

Auto: Enables automatic regulations of sites to Low rate encoder, based on video rate reports

Network Error Handling

None: Set to None to not enable error handling.

IPLR: Set to IPLR (Intelligent Packet Loss Recovery) if one or more sites are experiencing network errors.

FURBlock: Set to FURBlock (Fast Update Request Block) if one or more sites are experiencing network errors.

The Network Error Handling may be set to IPLR (Intelligent Packet Loss Recovery) or FURBlock (Fast Update Request Block) if one or more sites are experiencing network errors.

IPLR Robust Mode

Auto: When set to Auto, the IPLR Robust Mode is turned on for each encoder when needed.

On: When set to On, the IPLR Robust Mode is on for all encoders.

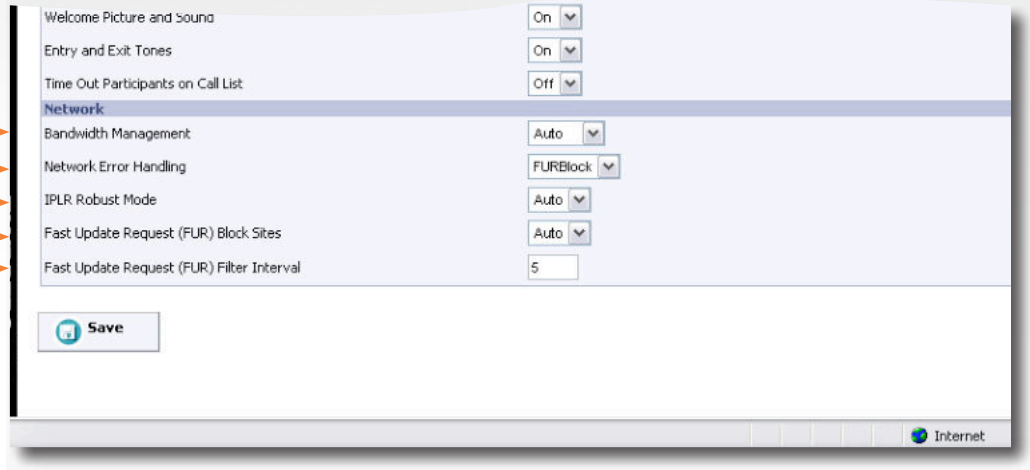
Please refer to Intelligent Packet Loss Recovery (IPLR) in the Appendices section.

Network

If a conference participant is experiencing poor network quality it will send Fast Update Requests (FUR) to the encoder in the MCU to make it refresh the picture. This can be observed as a short flash in the picture.

Poor network conditions for one participant may have a deteriorating effect on the video quality for some of the participants in the conference.

In an effort to reduce this effect the Network Error Handling can be used.



Press Save to activate the new settings

FUR Block Sites

Auto: When set to Auto, the FUR's from sites that send too many will be blocked.

On: When set to On, the FUR's from all sites will be blocked.

FUR Filter Interval

Denotes the number of seconds between FUR's, e.g. the minimum time between FUR's that will refresh the picture.

MCU Configuration

File Management

File Management

The File Management allows viewing and changing pictures, sounds and symbols, which are shown to the participants when connecting to, and during a conference on the MCU.

See also the Language submenu in System Configuration.

To add a new file, press Browse to find the file, and then press Upload. For each of the customized files, a Delete button will be added in the Type column.

Web Interface

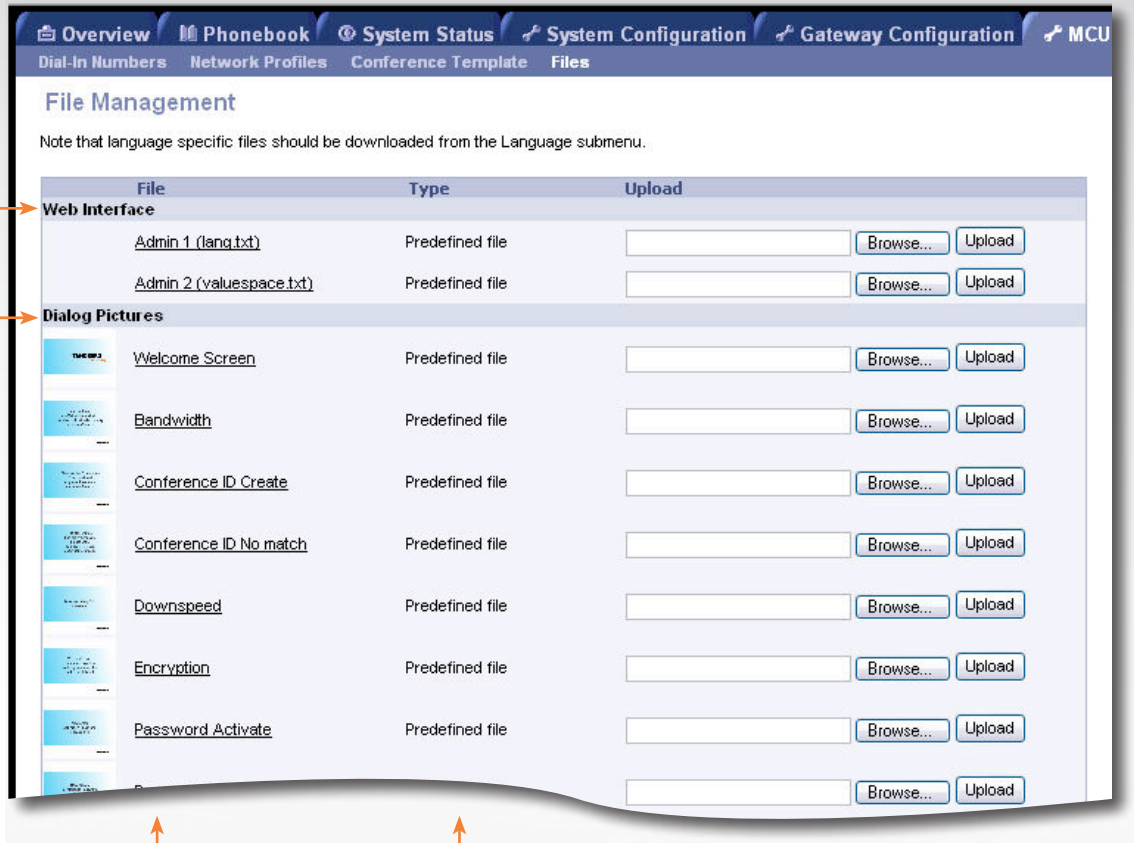
The following web interface options can be specified:

- Language - Upload language files
- Value Space - Upload Value Space files.

Dialog Pictures

Examples of MCU dialog pictures that can be specified:

- Welcome Screen
- Bandwidth
- Conference ID Create
- Conference ID Create No Match
- Downspeed
- Encryption
- Password Activate
- Password Create
- Password Enter
- Password Reject
- Ambigous Conference ID
- Now, Enter Conference ID
- Not Started Activate
- Not Started Wait
- Only Participant



File

Press the name of the File on the left side of the window to view or hear the content of the file.

Type

- When a customized file has been uploaded the text “Predefined file“ is replaced with a Delete button.
- When a customized file has been deleted the Delete button is replaced with the text “Predefined file“.

Browse

Press the Browse button to find a file.

Upload

Press the Upload button to upload the file.

MCU Configuration, *cont...*

File Management

File Management

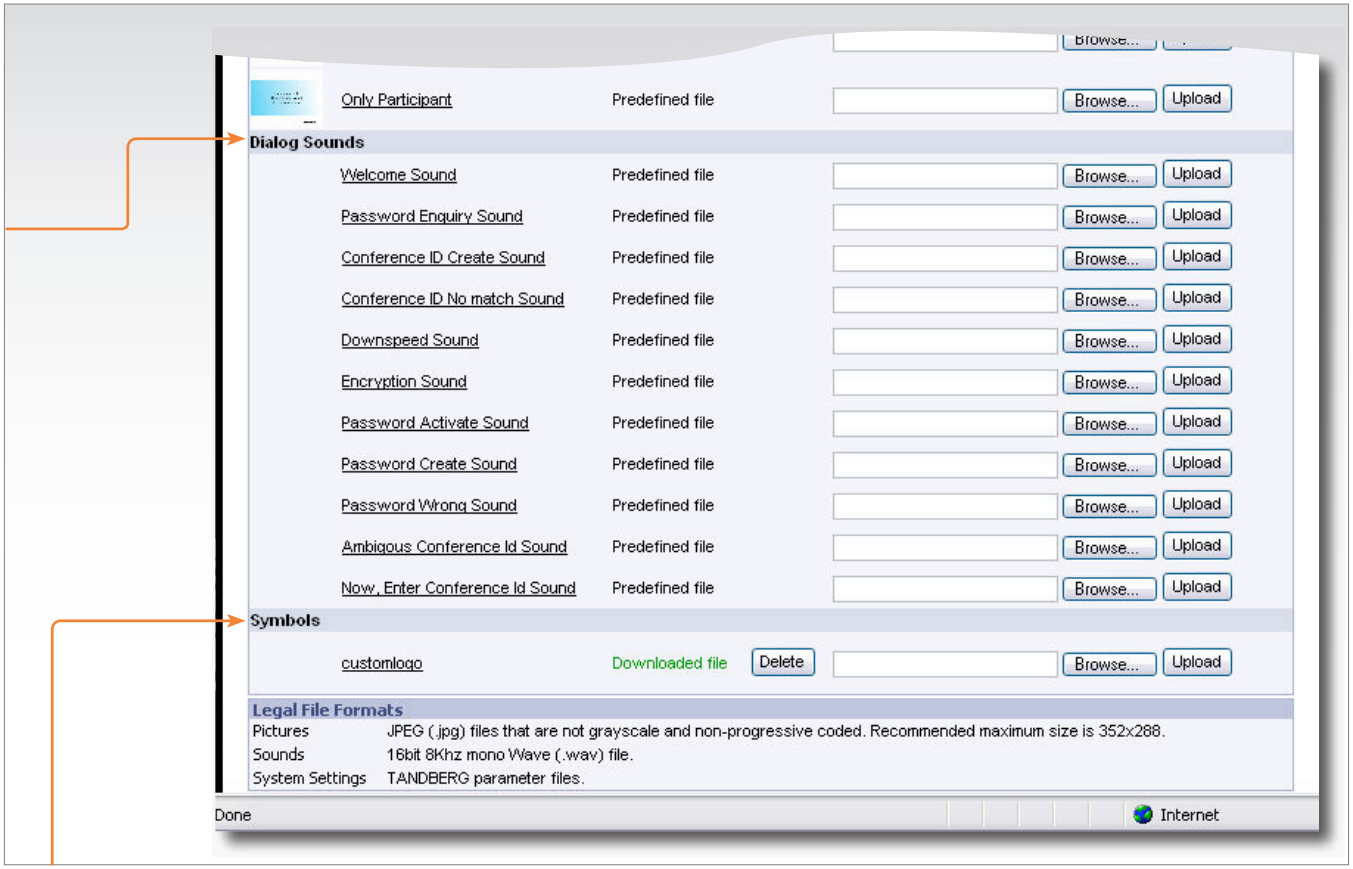
The File Management allows viewing and changing pictures, sounds and symbols, which are shown to the participants when connecting to, and during a conference on the MCU.

To add a new file, press Browse to find the file, and then press Upload. For each of the customized files, a Delete button will be added in the Type column.

Dialog Sounds

Examples of MCU dialog sounds that can be specified:

- Welcome Sound
- Password Enquiry Sound
- Conference ID Create Sound
- Conference ID Create No Match Sound
- Downspeed Sound
- Encryption Sound
- Password Activate Sound
- Password Create Sound
- Password Wrong Sound
- Ambiguous Conference ID Sound
- Now, Enter Conference ID Sound



Symbols

A Custom Logo can be specified:

- Recommended size is 64 x 64 pixels
- The MPS must be restarted to display the Custom Logo
- The Custom Logo will be presented in the upper left corners on all conferences hosted by the MPS. This logo can be used to identify/verify the MPS hosting the conference.

The logo will not be present in conferences which have enabled and are using optimal Voice Switch, oVS.

Legal File Formats

At the bottom of the page, custom file requirements are listed for each of the file-types.

Chapter 8

Appendices

The Appendices gives a detailed technical description of the System Controller Board, Media Boards and the different types of Network Interface Boards. You will also find more information on miscellaneous features on the MPS, how to insert additional network cards, system management and conference security etc.

You will also find the technical specification on the MPS 800 and MPS 200 along with other information you may need to better understand how to utilize the TANDBERG MPS.

Stay up-to-date

We recommend you visit the TANDBERG web site regularly for an updated version of this guide. Go to: http://www.tandberg.com/docs

In this chapter...

The System Controller Board Interface

TANDBERG MPS 800

The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis that can host up to 8 Media Processing Boards and 4 Network Interface Cards.

TANDBERG MPS 200

The TANDBERG MPS 200 has 3U-19" rack-mountable chassis that can host up to 2 Media Processing Boards and 2 Network Interface Cards.

Front Chassis

The TANDBERG MPS chassis is 19" rack-mountable. On the front of the chassis is a Liquid Crystal Display (LCD) for initial configuration and basic system information.

You will also find 4 Light Emitting Diodes (LEDs) for power status. The backplane of the chassis is provided with advanced CompactPCI technology for high speed communication between the boards. You will find 3 cooling fans in the lower front of the chassis.

Rear Chassis MPS 800

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units for configurations of 1 to 3 Media Processing Boards. If the unit has more than 3 Media Processing Boards the TANDBERG MPS 800 is equipped with 3 hot-swappable power units. The power units are installed at the back of the chassis. You will also find the power switch/connector at the back of the chassis.

Rear Chassis MPS 200

The TANDBERG MPS 200 is shipped with 1 power unit integrated in the chassis.

System Controller Board

The System Controller Board takes care of the following functions:

- Call control
- System management
- The embedded Web server

The LAN and Enet2 interfaces will allow you to connect to two different non-overlapping IP-networks so that participants with no IP-routing between them can be joined in the same conference. At least one Media Processing Board must be connected to each network.

The 2 x LAN interfaces will also give the TANDBERG MPS support for two Gatekeepers, one on each network.

Rear View

LAN Interface

The LAN interface on the System Controller Board is for management/call control signalling.

- 1 x LAN / Ethernet (RJ-45) 10/100 Mbps

USB Interface

USB0 and USB1 are for future use.

Serial Port Interface

To use the asynchronous serial port (J23) COM1 port, you need a RJ-45 to RS-232 converter. This port is configured as DCE.

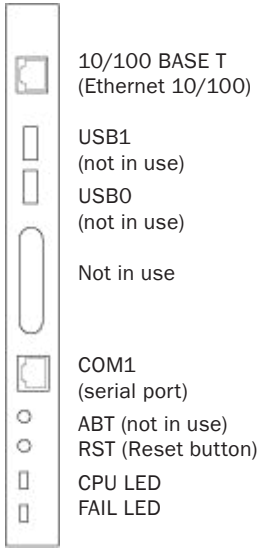
ABT and RST Buttons

RST - Press and hold RST button for a few seconds to restart the system. Restart can also be done via the LCD on the chassis.

CPU and FAIL LEDs

CPU LED - On when CPU activity

FAIL LED - On when parameter inconsistency in boot code



Rear View

Only the LAN interface Enet2 is in use. No other sockets on the rear System Controller Board are in use.

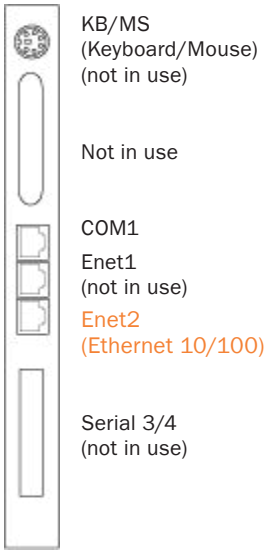
LAN Interface

The second LAN interface (Enet2) of the System Controller Board is accessible from the rear side.

The Enet2 interface is only in use if you are connecting the MPS to two separate IP networks.

Note that system management is disabled on Enet2. The Enet2 interface is only for call control.

- 2 x LAN / Ethernet (RJ-45) 10/100 Mbps (Enet1 not in use)



Serial Port Pinout

COM1 Pinout	
Pin	Description
1	DCD
2	RTS
3	GND
4	TXD
5	RXD
6	GND
7	CTS
8	DTR

The Media Processing Board Interface

TANDBERG MPS 800

The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis that can host up to 8 Media Processing Boards and 4 Network Interface Cards.

TANDBERG MPS 200

The TANDBERG MPS 200 has 3U-19" rack-mountable chassis that can host up to 2 Media Processing Boards and 2 Network Interface Cards.

Front Chassis

The TANDBERG MPS chassis is 19" rack-mountable. On the front of the chassis is a Liquid Crystal Display (LCD) for initial configuration and basic system information.

You will also find 4 Light Emitting Diodes (LEDs) for power status. The backplane of the chassis is provided with advanced CompactPCI technology for high speed communication between the boards. You will find 3 cooling fans in the lower front of the chassis.

Rear Chassis MPS 800

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units for configurations of 1 to 3 Media Processing Boards. If the unit has more than 3 Media Processing Boards the TANDBERG MPS 800 is equipped with 3 hot-swappable power units. The power units are installed at the back of the chassis. You will also find the power switch/connector at the back of the chassis.

Rear Chassis MPS 200

The TANDBERG MPS 200 is shipped with 1 power unit integrated in the chassis.

Media Processing Board

Add-on boards for media processing are installed in adjacent slots in the chassis.

The Media Processing Boards are handling the following functions:

- Video processing, see Video Features in the Appendices section for details.
- Audio processing, see Audio > Create Conference in the Using the MPS section for details.
- Transcoding, see Transcoding and Ratematching in the Appendices section for details.
- Encryption, see Secure Conference (Encryption) in the Appendices section for details.
- Continuous Presence/Voice Switching, see Video Features in the Appendices section for details.

TANDBERG MPS 800 has support for up to 8 Media Processing Boards.

TANDBERG MPS 200 has support for up to 2 Media Processing Boards.

Each Media Processing Board gives:

- 5 fully featured conferences
- 20 video sites @ 384kbps
- 16 audio sites @ 64kbps

One media board is, in conjunction with a network interface card, capable of a total of 7680 kbps for H.320/ISDN calls. For IP video calls without encryption it can handle 15360 kbps.

Two different IP Networks

If your TANDBERG MPS is connected to two different IP Networks utilizing both of the Ethernet ports (LAN and Enet2) on the System Controller Board, you need to pre-define the Media Processing Boards to Network #1 or Network #2. This is done in the Media Board Configuration in the System Configuration section, by choosing either Network ID 1 or 2 for the respective Media Processing Boards.

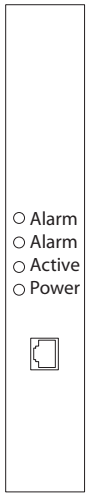
Front View

The Media Processing Board is equipped with LAN interface for H.323 media:

- 1xLAN/Ethernet (RJ-45) 10/100 Mbps

4 Light Emitting Diodes (LEDs) for board status.

1. Alarm: Red Alarm indicates that the Ethernet link is lost
2. Alarm: Flashes during startup and/or media processing board application failure.
3. Active: Green in normal operation
4. Power: Green in normal operation



4 LEDs

10/100 BASE T (Ethernet 10/100)

Advanced Video Option

Advanced Video Option

A Media Processing Board can be purchased with or without the Advanced Video Option (AVO). See Video Features in the Appendices section for further information.

All video participants will use one Advanced Video Option. In addition one Advanced Video Option will be used if Dual Stream is enabled in the conference.

The Advanced Video Option gives support for the following:

- Continuous Presence
- Dual Stream - support for both DuoVideo^{TF}, H.239 and BFCP
- Custom Video Formats (Digital Clarity^{TF})
- Best Impression^{TF}

All Media Boards should be connected to the IP network and be given an IP address to work properly, see Media Board IP Configuration for details.

E1/T1 Network Interface Card (IIC-8)

TANDBERG MPS 800

The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis that can host up to 8 Media Processing Boards and 4 Network Interface Cards.

TANDBERG MPS 200

The TANDBERG MPS 200 has 3U-19" rack-mountable chassis that can host up to 2 Media Processing Boards and 2 Network Interface Cards.

Front Chassis

The TANDBERG MPS chassis is 19" rack-mountable. On the front of the chassis is a Liquid Crystal Display (LCD) for initial configuration and basic system information.

You will also find 4 Light Emitting Diodes (LEDs) for power status. The backplane of the chassis is provided with advanced CompactPCI technology for high speed communication between the boards. You will find 3 cooling fans in the lower front of the chassis.

Rear Chassis MPS 800

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units for configurations of 1 to 3 Media Processing Boards. If the unit has more than 3 Media Processing Boards the TANDBERG MPS 800 is equipped with 3 hot-swappable power units. The power units are installed at the back of the chassis. You will also find the power switch/connector at the back of the chassis.

Rear Chassis MPS 200

The TANDBERG MPS 200 is shipped with 1 power unit integrated in the chassis.

E1/T1 Network Interface Card

- The MPS 800 supports up to 4 E1/T1 Network Interface Cards.
- The MPS 200 supports up to 2 E1/T1 Network Interface Cards.
- Each E1/T1 Interface Card gives 8 E1/T1 PRI ports.
- PRI ports/options are purchased in groups of 4 PRI ports.
- The E1/T1 Interface Card is also used for G.703 Leased Line calls.

ISDN Protocol

- All 8 PRI ports on the same PRI card must use the same ISDN protocol and the same network interface (E1 or T1).
- However, there is support for separate PRI protocols on each E1/T1 Interface Card.

Connecting the Card

- The E1/T1 Interface Card is installed in the rear of the chassis. See the TANDBERG MPS 800 – E1/T1 interface cards or TANDBERG MPS 200 – E1/T1 interface cards on how the card is installed.
- One Media Processing Board must be in front of each E1/T1 Interface Card.
- The Network Interface Card connects into the back of a Media Processing Board by connecting to its J5-connector from the back of the chassis.

Network Clock

The card connected to Media Processing Board #1 provides the primary network clock for the unit.

When you have an E1/T1 Interface Card connected to this Media Processing Board, the command line interface command reconfigures which PRI-port this clock is taken from:

```
xConfiguration SystemClock Port <0,1-8>
```

- The value 0 is auto, meaning the first port with a clock.
- The value 1 - 8 tells the TANDBERG MPS to take the clock from that specific PRI-port.

Rear View

The E1/T1 Card is equipped with PRI ports for ISDN interface:

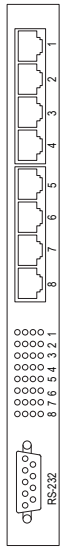
- 8 x E1/T1 PRI ports (RJ-45)

Light Emitting Diodes (LEDs) for PRI Port status, with 4 LEDs for each E1/T1 PRI port:

- Red – Layer 1 Red Alarm
- Yellow – Layer 1 Yellow Alarm
- Green – Layer 1 OK
- Red – D-Channel Down

See next page for more information about the LEDs.

Serial RS-232



8 x E1/T1 PRI Ports

4 LED's for each PRI Interface

Serial RS-232

PRI E1/T1 Cable

The cable of use should be a straight through configuration.

TANDBERG recommends always using at least category 5 cabling.

PRI E1/T1 Pinout	
PRI	Pinout
1	TIP RX
2	RING RX
4	RING TX
5	TIP TX

E1/T1 Network Interface Card (IIC-8), *cont...*

LED's for the E1/T1 PRI Interface Card (IIC-8)

You will find 4 Light Emitting Diodes (LEDs) for each PRI interface which provides information about PRI-line status.

Read the LEDs for each of the PRI interfaces on the E1/T1 Interface Card from left to right:

1. Red LED - If On this indicates Layer 1 Red Alarm
2. Yellow LED - If On this indicates Layer 1 Yellow Alarm
3. Green LED - If On this indicates Layer 1 OK
4. Red LED - If On this indicates that D-channel is Down

Red Alarm or Loss of Signal (LOS) indicates that there is no signal and thus no framing info received. The same effect will be obtained by pulling out the PRI cable. This may also be caused by a broken connector in the receive (RX) part of the cable.

Yellow Alarm or Remote Alarm Indicator (RAI) means that the MCU is receiving framing info, but in this framing info the other side tells the MCU that it is not reading the MCU's transmitted framing info. Typically, this may be a broken connector in the transmit (TX) part of the PRI cable. This could also indicate weak or noisy signal in the transmit (TX) part of the PRI cable.

Green LED or Layer 1 OK doubles as a clock source indicator. The LED for one of the eight interfaces will flash to indicate that the interface is being used as the clock source for H.110 bus.

- On - Layer 1 OK, not clock source
- On + Flash (long On/short Off) Layer 1 OK, used as clock source
- Off - Layer 1 Down, not clock source
- Off + Flash (long Off/short On) Layer 1 down, used as clock source

Red Alarm or D-Channel Down indicates that the ISDN out-of-band signalling channel is down. The D channel carries user-network signalling information and is primarily used in call setup and teardown. Note that for NFAS the D channel may not be used for all PRI interfaces. A red LED may thus not indicate anything wrong.

Channel Service Unit (CSU)

The PRI interface may require an external CSU (Channel Service Unit) depending on the network layout.

The Cable Length in the PRI configuration menu specifies the distance from the MCU to the CSU or last repeater.

Supported the PRI protocols

- AT&T Custom
- National ISDN
- Japan/Taiwan ISDN
- ETSI (Euro ISDN)

- The AT&T, Japan ISDN and National protocols gives a total of 23 B-channels per port
- The ETSI protocol gives a total of 30 B-channels per port.

Within these protocols the following switches are supported:

Switches Protocols supported

4ESS (AT&T)	AT&T Custom
5ESS (AT&T)	AT&T Custom and National ISDN
DMS250 (Nortel)	National ISDN
DMS100 (Nortel)	National ISDN
(Any switch) ETSI	(Euro ISDN)

Channel Hunting

Channel hunting is provided for outgoing calls. The feature is normally used when the number of channels needs to be specified.

- When no value is specified for low or high channel, they are default to 1 (low), 23 (high US) and 30 (high Europe).
- Default search is from high to low.

E1/T1 Network Interface Card (IIC-8), *cont...*

PRI T1 (US only)

Network Service Facility (NSF) can be configured to mode Off (i.e. NSF not used – default) or mode On with any value between 0 and 31, to describe the service facility on the PRI/T1 line.

ISDN Aggregation Standards

1. H.221 Frame Structure from 64 (56*) kbps to 128 kbps
2. ISO 13871 BONDING, Mode 1 from 64 (56*) kbps to 2 Mbps

H.221

- For bit rates from 56 kbps up to 128 kbps (1 or 2 channels).
- The maximum relative delay difference between the 2 B-channels is 0.6 second.

BONDING

- ISO 13871, BONDING Mode 1 for bit rates from 56 kbps up to 2 Mbps (1 to 30 channels).
- The maximum relative delay difference between B-channels is 0.5 second (i.e. to compensate for different routing of channels).

Standard bandwidths on H.320

30ch – 24 – 23 – 18 – 12 – 8 – 6 – 5 – 4 – 3 – 2 – 1

ISDN Channel Setup

The following is a description of how the ISDN channels are set up.

- Incoming & Outgoing MCU calls: Normally the MCU will set up only 1 channel from PRI 1 and build up the channels starting from the bottom of the last PRI in use.
- This will ensure that the MCU always have available channels on the first PRI number (which normally should be the MCUs main number).
- The TANDBERG MPS supports simultaneous dial in to the same conference on ISDN.

NSF Service codes

AT&T offers several digital switched services. These include SDN with service code 1 and ACCUNET with service code 6.

Below is a list of common service profiles. As these profiles may change, contact your service provider to get the correct profile.

AT&T Service Code (Ref.1)	
Code	Service
0	Disable *
1	SDN (including GSDN)
2	Toll Free Megacom (800)
3	Megacom
6	ACCUNET Switched Digital Service (incl. Switched Digital International)
7	Long Distance Service (incl. AT&T World Connect)
8	International Toll Free Service (1800)
16	AT&T MultiQuest
23	Call Redirection Service

Sprint Service Code (Ref.2)	
Code	Service
0	Reserved
1	Private
2	Inwatts
3	Outwatts
4	FX
5	TieTrunk

MCI Service Code (Ref.2)	
Code	Service
1	VNET / Vision
2	800
3	PRISM1, PRISMII, WATS
4	900
5	DAL

* “0” will still send NSF in the Q931 setup, which may cause calls to fail. Set to mode “off” if not needed.

Ref. 1: AT&T TR 41459 Specification, June 1999, page 76

Ref. 2: Ascend Multiband Plus-T1/PRI, User Documentation, Page 6-8

V.35 Serial Interface Card (SIC-32)

TANDBERG MPS 800

The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis that can host up to 8 Media Processing Boards and 4 Network Interface Cards.

TANDBERG MPS 200

The TANDBERG MPS 200 has 3U-19" rack-mountable chassis that can host up to 2 Media Processing Boards and 2 Network Interface Cards.

Front Chassis

The TANDBERG MPS chassis is 19" rack-mountable. On the front of the chassis is a Liquid Crystal Display (LCD) for initial configuration and basic system information.

You will also find 4 Light Emitting Diodes (LEDs) for power status. The backplane of the chassis is provided with advanced CompactPCI technology for high speed communication between the boards. You will find 3 cooling fans in the lower front of the chassis.

Rear Chassis MPS 800

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units for configurations of 1 to 3 Media Processing Boards. If the unit has more than 3 Media Processing Boards the TANDBERG MPS 800 is equipped with 3 hot-swappable power units. The power units are installed at the back of the chassis. You will also find the power switch/connector at the back of the chassis.

Rear Chassis MPS 200

The TANDBERG MPS 200 is shipped with 1 power unit integrated in the chassis.

V.35 Serial Interface Card (SIC-32)

- 32 x V.35 ports with optional RS366 Dialling
- Each port support rates from 64kbps up to 1920kbps
- Each port can either be dual-clocked (RS449, RS530, and V.35), or single clocked for X.21 applications.
- V.35 ports/options are purchased in groups of 8 V.35 ports.

Connecting the Card

A Media Processing Board must be in front of each V.35 Serial Interface Card. See the TANDBERG MPS 800 V.35 interface cards on how the V.35 Interface Cards are installed in the rear of the chassis.

V.35 is shipped as a kit. The V.35 kit includes the following:

- The V.35 Serial Interface Card
- 4 cables that convert from high-density connectors on V.35 card to TANDBERGs standard V.35 connectors (26pin DSUB)
- 19" rack-mountable panel where the V.35 connectors will fit.

Light Emitting Diodes (LEDs)

You will also find Light Emitting Diodes (LEDs) on the board. The LEDs gives you visual feedback on the status of the board.

Cable Specifications

To get more information on the cables needed to connect to the 26pin DSUB, refer to the following Cable Specification Documents:

- V.35 Cable specification, D1231201
- V.35 and RS366 Cable specification, D1230501
- X.21 Cable specification, D1230101

The above specification documents can be found on http://www.tandberg.com/support/documentation.php?p=Components_and_Cables

Rear View

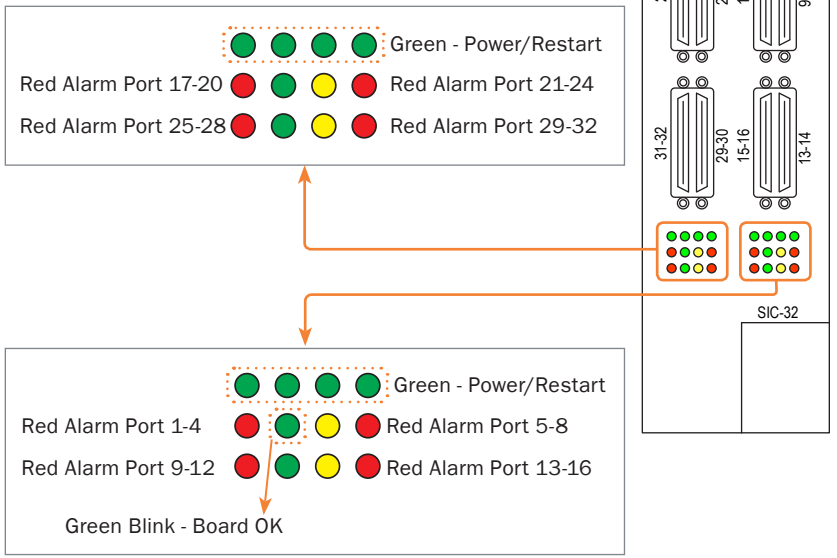
The V.35 Card is equipped with V.35 ports for serial interface:

- 32 x V.35 ports with optional RS366 Dialling

Light Emitting Diodes (LEDs) for V.35 Port status:

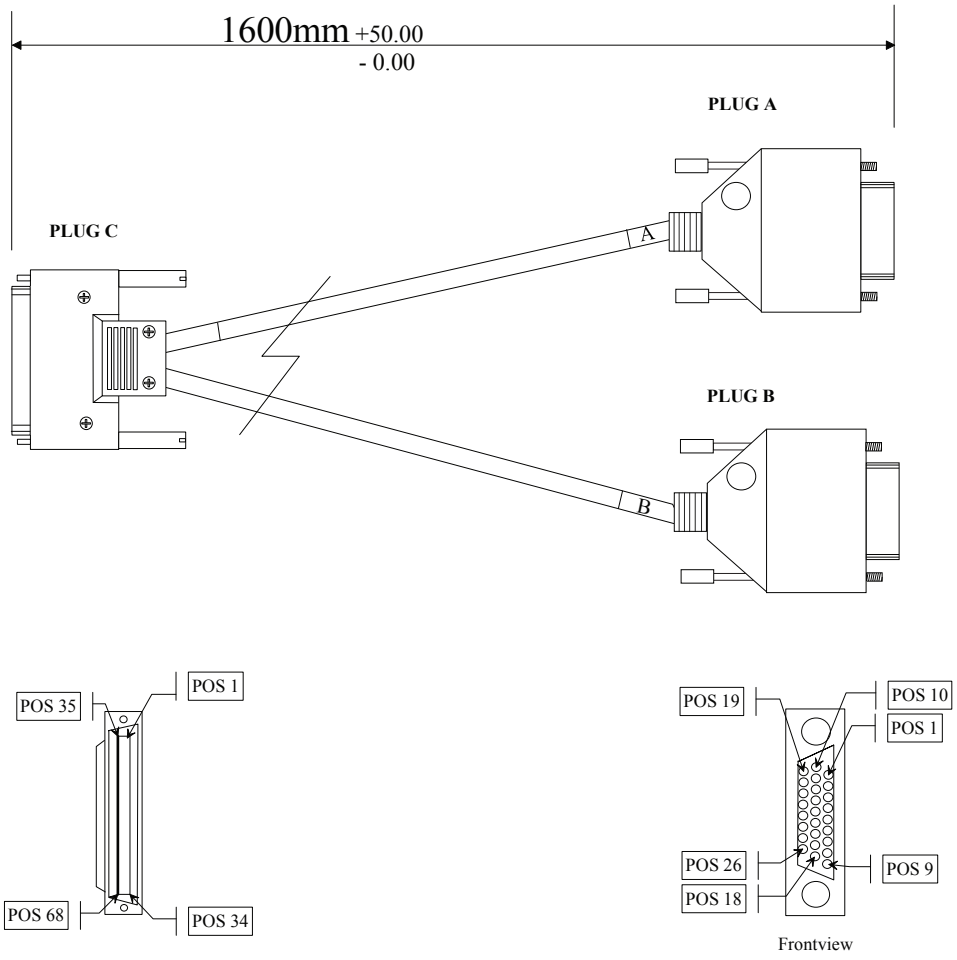
- Green Blink – The LED flashes every 2nd second to indicate the driver on the board is OK.
- Red Alarm - The LED(s) turns red when an alarm appear on any of the ports. Each LED indicates status of four ports.
- Green Power/Restart - The LEDs indicates the power status. While in restart mode one LED will blink.
- Yellow LEDs - Always On

For more information about status on the V.35 card, see System Status > Serial V.35 Board Status



V.35 Serial Interface Card (SIC-32), *cont...*

Cable Description



V.35 Pinout Table Plug A

Pair no	Plug C	Plug A	Signal Name
1	1	25	LOST SIG
1	2	NC	GND
2	3	18	CLK TX +
2	4	17	CLK TX -
3	5	12	TX +
3	6	11	TX -
4	7	14	RX +
4	8	13	RX -
5	9	19	GND
5	10	24	INCALL
6	11	21	TR +
6	12	20	TR -
7	13	16	CLK RX +
7	14	15	CLK RX -
8	15	23	RR +
8	16	22	RR -
9	17	26	DTR
9	18	NC	GND
10	19	7	RS_366 1
10	20	8	RS_366 2
11	21	9	RS_366 4
11	22	10	RS_366 8
12	23	NC	GND
12	24	2	DPR
13	25	3	ACR
13	26	4	CRQ
14	27	5	PND
14	28	6	DLO
15	29	NC	VCC
15	30	NC	VCC
16	31	1	GND
16	32	NC	GND
17	33	NC	GEN IN
17	34	NC	I_CALL OUT

V.35 Pinout Table Plug B

Pair no	Plug C	Plug B	Signal Name
1	35	25	LOST SIG
1	36	NC	GND
2	37	18	CLK TX +
2	38	17	CLK TX -
3	39	12	TX +
3	40	11	TX -
4	41	14	RX +
4	42	13	RX -
5	43	19	GND
5	44	24	INCALL
6	45	21	TR +
6	46	20	TR -
7	47	16	CLK RX +
7	48	15	CLK RX -
8	49	23	RR +
8	50	22	RR -
9	51	26	DTR
9	52	NC	GND
10	53	7	RS_366 1
10	54	8	RS_366 2
11	55	9	RS_366 4
11	56	10	RS_366 8
12	57	NC	GND
12	58	2	DPR
13	59	3	ACR
13	60	4	CRQ
14	61	5	PND
14	62	6	DLO
15	63	NC	VCC
15	64	NC	VCC
16	65	1	GND
16	66	NC	GND
17	67	NC	GEN IN
17	68	NC	I_CALL OUT

Power Supplies

TANDBERG MPS 800

The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis that can host up to 8 Media Processing Boards and 4 Network Interface Cards.

TANDBERG MPS 200

The TANDBERG MPS 200 has 3U-19" rack-mountable chassis that can host up to 2 Media Processing Boards and 2 Network Interface Cards.

Front Chassis

The TANDBERG MPS chassis is 19" rack-mountable. On the front of the chassis is a Liquid Crystal Display (LCD) for initial configuration and basic system information.

You will also find 4 Light Emitting Diodes (LEDs) for power status. The backplane of the chassis is provided with advanced CompactPCI technology for high speed communication between the boards. You will find 3 cooling fans in the lower front of the chassis.

Rear Chassis MPS 800

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units for configurations of 1 to 3 Media Processing Boards. If the unit has more than 3 Media Processing Boards the TANDBERG MPS 800 is equipped with 3 hot-swappable power units. The power units are installed at the back of the chassis. You will also find the power switch/connector at the back of the chassis.

Rear Chassis MPS 200

The TANDBERG MPS 200 is shipped with 1 power unit integrated in the chassis.

Power Supply for TANDBERG MPS 800

The TANDBERG MPS 800 is shipped with 2 hot-swappable power units. They are installed at the back of the chassis.

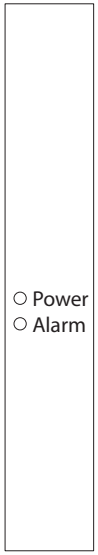
If the MPS has more than 3 Media Processing Boards the TANDBERG MPS is shipped with 3 hot-swappable power units.

The power switch/connector is found at the back of the chassis.

Light Emitting Diodes (LEDs)

On the rear side you will find 2 Light Emitting Diodes (LEDs) for power status on each of the power supplies:

- Power: Green when in in normal operation
- Alarm: Red indicates power failure



MPS 800–Inserting additional network interface cards

TANDBERG MPS 800

The TANDBERG MPS 800 has a 9U-19" rack-mountable chassis.

- There is support for up to 4 Network Interface Cards in the software version J3 and above.
- There can be a mix of Network Interface Cards
- Network Interface Cards are installed in the rear side of the chassis.
- One Media Processing Board must be in front of each Network Interface Card.

PRI E1/T1 ISDN Interface Card

You can install up to 4 PRI E1/T1 ISDN Interface Cards.

- The PRI E1/T1 ISDN Interface Cards are installed in the rear of the chassis.
- One Media Processing Board must be in front of each ISDN Interface Card.

V.35 Serial Interface Card

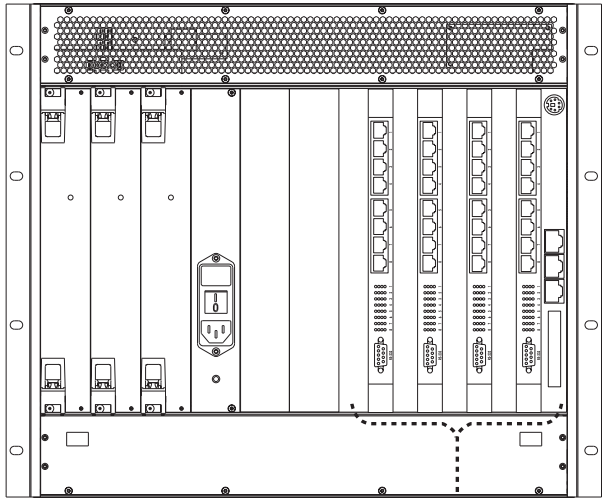
You can install up to 4 V.35 Serial Interface Cards.

- The V.35 Interface Cards are installed in the rear of the chassis.
- One Media Processing Board must be in front of each V.35 Interface Card.

Mix of Network Interface Cards

You can install a mix of PRI E1/T1 ISDN Interface Cards and V.35 Serial Interface Cards.

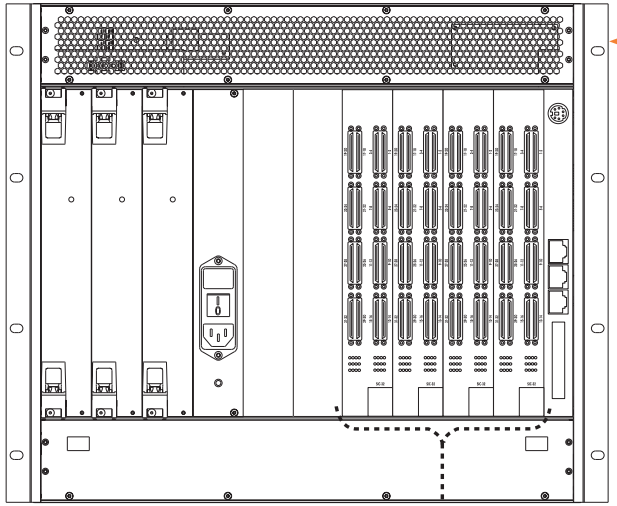
- The PRI E1/T1 ISDN Interface Cards and the V.35 Interface Cards are installed in the rear of the chassis.
- One Media Processing Board must be in front of each ISDN/V.35 Interface Card.



PRI E1/T1 ISDN Interface Card

4 x PRI E1/T1 ISDN Network Interface Cards on rear side

4 x Media Processing Cards in front



V.35 Serial Interface Card

4 x V.35 Serial Interface Cards on rear side

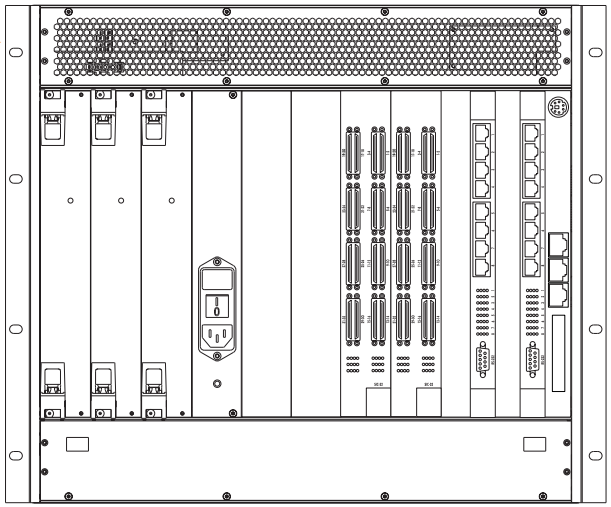
4 x Media Processing Cards in front

Mix of Network Interface Cards

2 x PRI E1/T1 ISDN Network Interface Cards on rear side

2 x V.35 Serial Interface Cards on rear side

4 x Media Processing Cards in front



MPS 200–Inserting additional network interface cards

TANDBERG MPS 200

The TANDBERG MPS 200 has 3U-19" rack-mountable chassis.

- There is support for up to 2 Network Interface Cards in the software version J3 and above.
- There can be a mix of Network Interface Cards
- Network Interface Cards are installed in the rear side of the chassis.
- One Media Processing Board must be in front of each Network Interface Card.

PRI E1/T1 ISDN Interface Card

You can install up to 2 PRI E1/T1 ISDN Interface Cards.

- The PRI E1/T1 ISDN Interface Cards are installed in the rear of the chassis.
- One Media Processing Board must be in front of each ISDN Interface Card.

V.35 Serial Interface Card

You can install up to 2 V.35 Serial Interface Cards.

- The V.35 Interface Cards are installed in the rear of the chassis.
- One Media Processing Board must be in front of each V.35 Interface Card.

Mix of Network Interface Cards

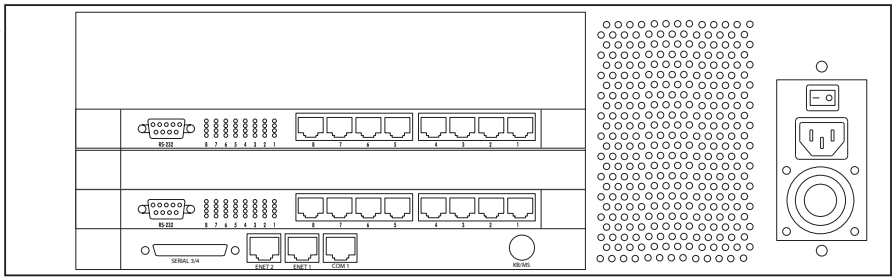
You can install a mix of PRI E1/T1 ISDN Interface Cards and V.35 Serial Interface Cards.

- The PRI E1/T1 ISDN Interface Card and the V.35 Interface Card are installed in the rear of the chassis.
- One Media Processing Board must be in front of each ISDN/V.35 Interface Card.

PRI E1/T1 ISDN Interface Card

2 x PRI E1/T1 ISDN Network Interface Cards on rear side

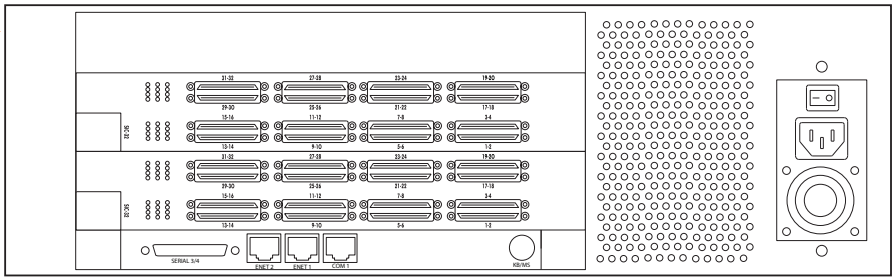
2 x Media Processing Cards in front



V.35 Serial Interface Card

2 x V.35 Serial Interface Cards on rear side

2 x Media Processing Cards in front

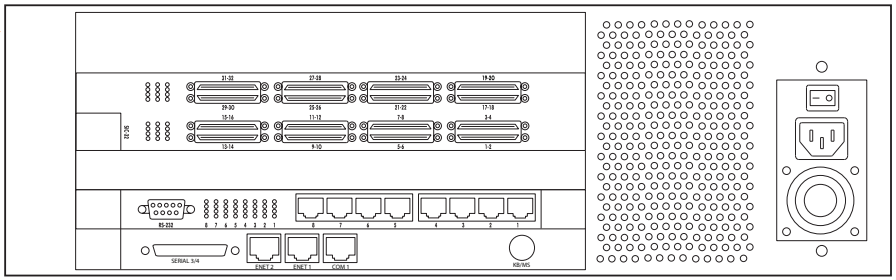


Mix of Network Interface Cards

1 x PRI E1/T1 ISDN Network Interface Cards on rear side

1 x V.35 Serial Interface Cards on rear side

2 x Media Processing Cards in front



About the Quality of Service (QoS) feature

Quality of Service (QoS)

All QoS features must be enabled in your IP network in addition to correct MPS configuration.

QoS Configuration for the MPS is done from System Configuration in the web interface, see System Configuration > QoS

DiffServ

DiffServ defines which priority audio, video, data and signalling packets should have in an IP network.

IP Precedence

IP Precedence defines which priority audio, video, data and signalling should have in an IP network. The higher the number, the higher the priority.

Type of Service

Type of Service enables the user to define what type of connection that should be chosen for the IP traffic. Helps a router select a routing path when multiple paths are available.

IP Precedence

IP Precedence is a classification of packets from 0 (low priority) to 7 (high priority). The values 6 and 7 are typically reserved for congestion control. IP Precedence helps a router select what kind of traffic to prioritize. By means of queue mechanisms, it can select which packets to send first and which to throw away. Some information/traffic is time critical while other is not, and classification is used to differentiate this traffic.

One may set separate IP Precedence for Signalling, Audio, Video and Data (values 1 – 7) as well as turn IP Precedence off.

The Auto setting uses the following values for IP Precedence:

- Signalling = 4
- Audio/Video = 4
- Data = 4 (e.g. FECC commands)

Separate priorities for Audio and Video are not recommended as this may cause reduced quality and lipsync problems.

Differential Services (DiffServ)

Differential services is another method of QoS offered by TANDBERG that utilizes 6 bits of the Type of Services Byte. Differential services is an extension of IP Precedence, where one can set values from 0 to 63 (63=Highest priority). This method is currently replacing IP Precedence as the preferred method for setting priority of packet traffic.

IP Type of Service (TOS)

TOS helps a router select a routing path when multiple paths are available.

- Delay - tells router to minimize delay
- Throughput - tells router to maximize throughput
- Reliability - tells router to maximize reliability
- Cost - tells router to minimized cost
- Off - Turns TOS off

About video features

TANDBERG Best Impression[TF]

TANDBERG's Best Impression[TF] feature consists of 3 parts:

- Automatic Continuous Presence (CP) layout switching
- Automatic Continuous Presence (CP) resolution switching
- Enhanced video transcoding

Automatic Continuous Presence (CP) Layout Switching

When Conference Layout is set to Auto the following table describes the picture sent from the TANDBERG MPS.

Layout Switching	
No of video sites in conference	Picture Mode used
1 or 2	Full Screen
3 -> 6	5 + 1 split
> 6	7 + 1 split

The TANDBERG MPS allows for changing between any of these modes during the conference.

Automatic Continuous Presence (CP) Resolution Switching Voice Switched Mode

In Voice Switched mode or when someone has the floor the TANDBERG MPS will send the same format as the one received if all sites are capable of receiving it.

If one of the sites cannot receive the preferred video format (or if a site asks for a video format not supported by the MCU), the MCU will fallback according to the following table:

Voice Switched Mode	
Preferred Video Format	Fallback Video Format
XGA	SVGA
SVGA	VGA
VGA	4CIF
4SIF	4CIF
4CIF	H.263 CIF
H.264 SIF	H.264 CIF
H.264 CIF	H.263 CIF
H.263 SIF	H.263 CIF
H.263 CIF	H.261 CIF
H.261 CIF	H.263 QCIF
H.263 QCIF	H.261 QCIF

Automatic Continuous Presence (CP) Resolution Switching DuoVideo[TF]/H.239/BFCP

DuoVideo[TF]/H.239/BFCP may be sent in both Continous Precence and Voice Switched mode.

- The TANDBERG MPS will transmit the same format as the one received if all the sites are capable of receiving it.
- The Video Format will fallback is as described for Voice Switched mode.

Automatic Continuous Presence (CP) Resolution Switching Continuous Presence Mode (CP)

In Continuous Presence mode the TANDBERG MPS administrator may set the conference up for Motion, Sharpness or Auto:

Continuous Presence Mode			
Video Format Modes	CP Mode	Meeting Rate	MCU will transmit the highest possible common video format according to:
MOTION	Any	Any	H.263 w288p -> H.264 CIF -> H.263 CIF -> H.261 CIF -> H.263 QCIF -> H.261 QCIF
SHARPNESS	Any	Any	H.263 w576p -> H.263 4CIF -> H.264 CIF -> H.263 CIF -> H.261 CIF -> H.263 QCIF -> H.261 QCIF
AUTO (4:3)	Any	< 256 kbps	Same formats as for MOTION, excluding w288p
AUTO (16:9)	Any	< 512 kbps	w288p
AUTO (4:3)	Any	=> 256 kbps	Same formats as for SHARPNESS, excluding w576p
AUTO (16:9)	Any	=> 512 kbps	w576p

About video features, *cont...*

4CIF Digital Clarity[TF]

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

H.263+ Custom Video Formats (Digital Clarity[TF])

TANDBERG MPS supports encoding of a true XGA, SVGA, VGA and NTSC (SIF) resolutions.

- This allows the TANDBERG MPS to send all video signals in their true native resolution.
- If the source is sending XGA, SVGA or VGA, the receiving system will decode and display the same quality image as was sent by the transmitting side.
- Also supported under the H.263+ custom formats is TANDBERG's use of SIF (Source Input Format) that allows for an improved NTSC image over traditional CIF based MCU.

H.263+ Widescreen formats

TANDBERG MPS support encoding of true widescreen (16:9) formats: w288p (512x288) and w576p (1024x576).

- Widescreen formats will only be used if all participants in the conference support it.
- When encoding widescreen formats, the TANDBERG MPS will use special layouts designed to utilize widescreen monitors.

See Conference Layouts in the Create Conference view for a list of all available layouts.

Dual Video Stream - DuoVideo[TF] / H.239 / BFCP

Dual Video Stream (DuoVideo[TF]/H.239/BFCP) is a feature that allows two simultaneous video streams to be transmitted from one system and be received by the other participants. This feature is supported on both H.320, H.323 and SIP. This allows a presenter and their presentation to be seen at the same time, in much the same way you would expect them to be in a local room environment.

If systems in a conference are TANDBERG dual monitor systems, they will receive the presenter on one screen and the presentation (PC, DocCam, VCR, DVD etc) on the other.

If the TANDBERG system is a single monitor system with default settings, the video stream will automatically switch from the presenter to the presentation. This change will take place on the fly, with the displayed stream returning to the presenter when the Dual Video Stream is disconnected.

In the TANDBERG MPS there is limited support for a mix of DuoVideo[TF] and H.239/BFCP in the same conference. Other manufacturer's systems supporting neither DuoVideo[TF] nor H.239/BFCP can be in the same conference when the second video stream is sent, but they will not be able to receive the second stream. In a mixed conference, it is not possible to take over floor without releasing the current floor holder first.

The TANDBERG MPS is capable of receiving/transmitting two simultaneous video signals from/to all sites.

1. When one of the sites requests for Dual Video the TANDBERG MPS will open Dual Video towards all the other video sites capable of receiving it.
2. Now, if another site requests for Dual Video the TANDBERG MPS will automatically transmit this new site's Dual Video to the rest of the sites if possible.

If H.239 or BFCP is not supported by a H.320 site, the TANDBERG MPS uses High Speed Data rates (HSD) for DuoVideo[TF]. If any of these sites does not support HSD, DuoVideo[TF] will still be transmitted to the sites supporting this. The sites not supporting HSD will in that case receive the same video stream as the others see on their main video (and therefore not use all of their available meeting rate).

Dual Video Streams are available in both Continuous Presence mode (CP) and Voice Switched mode or when one site has the floor, regardless of which site is broadcasted on the main stream.

Note that when running DuoVideo[TF] the video algorithm for the main stream will not be H.264.

Automatic Bandwidth Adjustment

When Dual Video is sent to the sites, the TANDBERG MPS will automatically use approximately 1/2 of the original video bandwidth for the main video stream and the other half for Dual Video.

When Dual Video is no longer transmitted, the TANDBERG MPS will automatically upspeed the main video stream to its original bandwidth.

The ITU standard H.264

The ITU standard H.264 provides considerably better video quality at lower bandwidths.

It has been developed with strong TANDBERG participation in joint workgroups of ITU-T and ISO.

- H.264 is based on the ISO standard MPEG-4.
- The TANDBERG MPS supports H.264 video compression in both CIF and SIF resolutions.
- An encoding and decoding rate of a high-quality 30 frames per second is achieved.
- H.264 may be used on the TANDBERG MPS for endpoint up to 768kbps.

Optimal Voice Switch

In Optimal Voice Switched mode the MPS will automatically send the optimal video quality to all participants capable of receiving the video quality of the active speaker.

- To get the optimal quality video, no icons or text are added to the video in this mode.
- If there is a site on the conference connected on H.320 special means are taken to ensure optimal interworking.
- Sites on H.320 will receive transcoded video while sites on IP will receive switched video.

About other features

IP Adaptive Bandwidth Management

The MCU never produces more traffic than needed, for better utilization of network resources. Most of the data sent in a videoconference is video data. Thus, by incorporating smart video algorithms, the codec sends no more video data than necessary. Little movement in the picture gives low bit rate; while a lot of movement gives higher bit rate.

The MCU regulates outgoing and incoming media bit rates by means of flow control signalling.

An example of this is automatic adjustment of total bandwidth used when DuoVideo[TF] is opened.

Asymmetrical Encoders and Decoders

Realizing there are many different types of videoconferencing units in the world that do not support the same video and audio algorithms, TANDBERG has implemented asymmetrical encoders and decoders.

This feature allows different systems with different video and audio algorithms to communicate with the TANDBERG MPS without having to settle on the best common protocols.

The asymmetrical nature of the TANDBERG MPS allows the MCU to accept any of its supported algorithms in any combination from any site and will always transmit the highest possible quality video and audio to the far end.

By this, the TANDBERG MPS can send H.263 and receive H.264 at the same time.

Lip Sync

IP is an asynchronous network that sends audio and video separately. It is easy to see that these two streams may not arrive at their destination at the same time. Lip sync problems are a certainty if precautions are not taken in the implementation of the codec.

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

Latency & Jitter

Latency is defined as the time between a node sending a message and receipt of the message by another node.

TANDBERG MPS can handle any value of latency however, the higher the latency, the longer the delay in video and audio. This may lead to conferences with undesirable delays causing participants to interrupt and speak simultaneously.

Jitter is defined as the difference in latency. Where constant latency simply produces delays in audio and video, jitter can have a more adverse effect. Jitter causes packets to arrive out of order or at the wrong times, which again leads to packet loss.

TANDBERG MPS can manage packets with jitter up to 200ms. If excessive packet loss is detected, the TANDBERG MPS will downspeed the connection until acceptable packet loss is achieved.

Flow Control (Downspeeding[TF])

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

If the ISDN networks drops channels during a call, the conference will not shut down but adjust to the remaining number of available channels. This ability is called “Downspeeding” and is in accordance to the BONDING Mode 1 standard.

ISDN Downspeeding will be started when one of the following actions occurs:

1. Fallback to 2xH.221 call when no BONDING framing is found
2. Non-matching number of channels (N) during BONDING setup (e.g. N=4 on site A, N=6 on site B)
3. Downspeeding during call setup caused by a channel that will not connect
4. Disconnect and downspeed if a channel will not sync
5. Downspeeding when detecting B-channels that are looped
6. Downspeeding if a channel drops during the call

The TANDBERG system will start the BONDING synchronization procedure if there is a long fatal failure situation:

7. Extended loss of H.221 framing
8. Continuous BCH framing error in the video stream. If then a channel fails to sync, it will be disconnected (as mentioned in 4.)

Packet Loss

This feature comes also in handy when using H.323 over networks with poor QoS such as the Internet.

If TANDBERG MPS detects excessive packet loss, it will use Flow Control to downspeed the far end to overcome the packet loss problem.

Packet loss can occur when routers become overloaded and discard packets or when the receiving video system cannot keep up with the transmitting video system.

Downspeeding on IP

The TANDBERG system follows a multilevel intelligent adaptive algorithm based on measured packet loss when downspeeding on IP.

Different from the ISDN downspeeding, the system will not change Audio and Video algorithms during the downspeeding of an IP call.

About other features, *cont...*

Intelligent Packet Loss Recovery (IPLR[TF])

IPLR is an ITU standard based packet loss compensation for H.323 that improves received video into the TANDBERG MPS. IPLR supports all video protocols and resolutions that TANDBERG MPS already has implemented and is compatible with all terminals and Gateways. This method works with other vendors' endpoints as well as MCUs. Video quality is improved on the transmitted and received video streams.

IPLR is a special algorithm developed at TANDBERG that will make efforts to reconstruct the lost packets and reduce the visual effects caused by packet losses.

If the TANDBERG MPS experiences packet loss from an endpoint, it will ask the endpoint to handle packet loss. This requires Intelligent Packet Loss Recovery functionality on the endpoint.

See White Paper TANDBERG and Packet Loss on http://www.tandberg.com/docs for details on the IPLR[TF] feature.

Inband Changing of Video and Audio Algorithms

TANDBERG MPS is able to change its video and audio algorithms 'on the fly' during a conference. This is valuable when switching between the live user image (CIF) and the live PC (XGA, SVGA, VGA).

Transcoding and Ratematching

Audio:

Transcoding: Each of the sites is negotiating the audio algorithm individually. So in a Multipoint call we support any combination of all the supported audio codecs (G.711, G.722, G.722.1, G.728, MPEG-4 AAC-LD). The same goes for ratematching, any combination is supported.

Video:

All sites are divided into groups of similar coding (H.261, H.263 and H.264) and rate capabilities. The number of groups and their properties depends on the sites connected to the conference. During a conference call, group properties might change, and also sites might move between groups.

Distributed MCUs

Cascaded MPS

The TANDBERG MPS supports two-level cascading, so the TANDBERG MPS can be cascaded with other TANDBERG MPSs to increase the number of participants in one conference.

By simply dialling from one TANDBERG MPS to other TANDBERG MPSs one will achieve a distributed setup.

In this case the TANDBERG MPS dialling out will be defined as the master MCU and all other MCUs as slaves.

The connection between the MCUs (or the Endpoints) can be IP, ISDN or V.35.

The slave MCUs will automatically be set to Voice Switched while the master MCU may be set to one of the CP layouts or kept in Voice Switched mode.

A distributed MCU scenario may add delays to the switching of conference.

Features supported within a distributed MCU scenario:

- DuoVideo^{TF} from any site
- Encryption (as long as all sites support this feature)
- Request Floor from any site
- View Site naming (of any site in the conference)

Administrator features (on TANDBERG MPS or TMS):

- View site names (terminal list) of all participants
- Configure the conference as Master, Slave or Auto
  - Auto – The MCU dialling out will become Master
  - Master/Slave status on Web
- Conference Layout disabled on Slave MCU
- Assign Floor from Master MCU
  - Shows the site in full screen for all participants
- Assign Floor from Slave MCU
  - Shows the site in full screen for all participants
  - Only to sites on this MCU

Ports and Packet Sizes

Layer 4 Ports Used in H.323 Meetings

TCP and UDP Ports

The following TCP and UDP ports are relevant for the TANDBERG MPS.

TCP and UDP Ports for TANDBERG MPS			
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 - 6951	UDP	<=>
Audio	Range 2326 - 6951	UDP	<=>
Data / FECC	Range 2326 - 6951	UDP	<=>
SSH	22	TCP*	<=>
Telnet	23	TCP*	<=>
HTTP	80	TCP*	<=>
HTTPS	443	TCP	<=>
SNMP (Queries)	161	UDP	<=>
SNMP (Traps)	162	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. until it reaches 6555. It will then start on 5555 again, unless the TANDBERG MPS has been restarted in the meantime.

Ports for TMS

Ports TMS (TANDBERG Management Suite) uses for communicating with the TANDBERG MPS

Ports TMS uses with TANDBERG MPS		
Port	Functions	Protocol
443	Read system info/status Give dial commands Get call log	HTTPS
80	Events like boot, disconnect etc	HTTP traps to TMS
161 UDP	Check system status	SNMP Commands

If HTTPS (port 443) is not available on the endpoint, HTTP (port 80) will be used instead

IP Packet Sizes

Audio: The TANDBERG MPS is sending maximum 320 bytes of audio per packet.

Video: The TANDBERG MPS is sending maximum 1400 bytes of video per packet.

In addition, the system needs to add the following header information (40 bytes in total) for each of the audio and video packets above:

- 20 bytes IP-header
- 8 bytes UDP-header
- 12 bytes RTP-header

Packet loss: Packet loss is displayed in the conference overview of the TANDBERG MPS, see MCU Conference Overview. The number shown is the average of audio- and video- packets that are lost during the last 2 seconds.

Secure Conference (Encryption)

Built-In Encryption

The TANDBERG MPS has built-in encryption of audio, video and data for:

- H.323 meetings (based on ITU standard H.235 v2&v3)
- H.320 meetings (based on ITU standard H.233 and H.234)

NOTE: Encrypted SIP calls is a Beta feature, and you should not run with Encrypted SIP calls in a production environment, it has limited support and is for evaluation and testing use only.

Encryption Mode

The administrator decides, when setting up the conference, whether or not a conference shall be in encrypted mode or in unencrypted mode. It is not possible to change the mode when the conference is active with calls.

Encryption Algorithm

The encryption algorithms used in the TANDBERG system are:

- The Data Encryption Standard (DES) with a 56 bits session key
- The Advanced Encryption Standard (AES) with a 128 bits session key

Typical Setup of an Encrypted Call

Although there are some differences between H.323 and H.320, a typical set-up of an encrypted call can be defined as follow:

1. Establishment of a common secret key and selection of an encryption algorithm.
2. Exchange of keys according to the common secret key and the selected encryption algorithm.
3. Start the encryption.

Common Secret Key

The establishment of the common secret key is done through the computation of a Diffie-Hellman (DH) algorithm. The DH method uses primes numbers of 512 bits length for DES and 1024 bits for AES.

Shared Secret Key

The shared secret key is then used as a key for the selected encryption algorithm, which encrypts the session keys. When the session key is collected by the remote end, encryption of the audio, video and data channels can start.

Establishment of Encryption

The encryption will be established automatically when all endpoints in the conference supports encryption, with automatic key generation (and the conference is set up for encryption mode of operation).

Encryption Support

Encryption is supported on all bandwidths. Encryption is also supported for DuoVideo^{TF} and H.239.

For an encrypted conference, all endpoints must support AES or DES encryption.

Encryption Configuration

Encryption is configured when you create a conference or a conference template from the web interface, see MCU Overview > Encryption Mode

- If Encryption Mode is set to Auto, the TANDBERG MPS accepts both AES and DES encryption.
- If Encryption Mode is set to AES, all participants must have AES in order to join the conference.

Scenarios

- If a site entering an encrypted conference does not support encryption, a picture will be shown, informing that the conference requires encryption.
- If a site connected to an encrypted conference starts sending unencrypted data, that site will be taken out of the conference.
- If the TANDBERG MPS administrator has forced the MCU to require only e.g. AES encryption then, all participants must have AES in order to join the conference.

For more information on AES and DES please visit the National Institute of Standards and Technology at http://www.nist.gov.

For more details see the White Paper TANDBERG MCU and IP and the White Paper TANDBERG Security documents on http://www.tandberg.com/docs.

System Management Tools

System Management Tools

TANDBERG provides a comprehensive set of system management tools, and is committed to the use of standards-based tools.

The TANDBERG MPS can be managed in many ways:

- The internal web server allows for remote control of the MCU using a web interface from anywhere on the LAN/WAN/Internet (open browser http://System Controller Board-ipaddress).
- SSH and Telnet for local control and diagnostics.
- The TANDBERG Management Suite (TMS) may also be used to manage the system and the TANDBERG Scheduler may be used to schedule conference meetings (Optional).
- XML and SOAP interface for full management of MPS through remote application.

Management using a standard Web-browser

The web-browser is the most common way to manage the TANDBERG MPS giving access to all managing features of the MCU.

Using a standard Web-browser (Mozilla Firefox, MS Internet Explorer 5.0 or later), the user may perform all forms of meeting set-up and control, but also diagnostics, troubleshooting and software upgrade.

Open a web browser and enter:

- `<http://MPS IP Address>` or
- `<https://MPS IP Address>`
- Example: `<http://10.0.5.203>`

Management using a SSH Client

The TANDBERG MPS also supports remote login through Secure Shell (SSH).This gives the user the same functionality as from the web interface. In addition it provides the user with advanced debug capabilities. Both ends of the client/server connection are authenticated using a digital certificate, and passwords are protected by being encrypted.

This interface is a command-line type interface, not a graphical interface like the Web-browser interface. The system supports multiple simultaneous SSH sessions.

See the TANDBERG MPS API guide for details on the API commands available via SSH.

Management using XML

System Management by using XML can be done from System Configuration > XML Document. The XML Uploading page, allows administrators to upload configuration changes to the MPS in one go, instead of manually setting each entry via the web interface.

Management using SOAP

By using the MPS WSDL file, developers can manage the MPS through SOAP web services. The wsdl file can be accessed on the MPS through a standard web browser at

`<http://<MPS IP Address/webservices.wsdl>`

These web services reflect the same functionality as can be found in the document:

`<http://MPS IP Address/command.xml>`

Management using TMS

Management of the MPS can also be achieved by registering the MPS with TMS (TANDBERG Management Suite).

Once registered to TMS, administrators will be able to update many of the MPS settings.

TMS users will also have the ability to automatically schedule conferences on the MPS using the settings defined in TMS.

Backup & Restore

From the TMS (TANDBERG Management Suite) administrators can backup and restore the configuration.

Management using a standard Telnet Client

This gives the user the same functionality as SSH and provides the user with advanced debug capabilities. The interface is a command-line type interface.

The system supports multiple simultaneous Telnet sessions.

Read about the API commands in the TANDBERG MPS API Guide. Go to: http://www.tandberg.com/docs.

Using SCP for Personal Conferences and Gateway Rules backup.

1. Login to the MPS using an SCP program such as WinSCP (Note that SSH must be enabled on the MPS to use SCP).
2. The user name will be "`root`" and the password will be the normal MPS password (default is TANDBERG).
3. Navigate to the "`persistent`" folder.
   - For Personal Conferences copy the `<conferencedefinitions.xml>` file
   - For Gateway rules copy the `<mappingrules.xml>` file

System Management and Security

HTTPS

The TANDBERG MPS supports HTTPS in order to ensure secure transmission of the information displayed on the administrator's PC.

A secure connection between the Web browser and the TANDBERG MPS (the MCU web server) will be established if the HTTPS service on the MCU is enabled.

HTTPS allows for password exchange, which is especially important.

Enable HTTPS using API Command

1. To enable HTTPS, you can use the API command service:
   `<xconfiguration HTTPS Mode: On>`
2. Restart: The HTTPS service will be activated at next restart.

Enable HTTPS using Web Interface

1. To enable HTTPS, navigate to System Configuration > Misc > Services and set HTTPS to On.
2. Press Save button to save the changes.
3. Press Restart button. The HTTPS service will be activated at next restart.

HTTP and HTTPS Service

If both the MCUs HTTP and HTTPS services are activated, the user will automatically be redirected to HTTPS.

If HTTP is deactivated, you will have to specify HTTPS. In this case https://10.0.5.203 will work, but not http://10.0.5.203.

Disable Services

For security reasons you may want to disable some of the services provided by the TANDBERG MPS. If wanted the following services may be disabled/enabled (either through the web-interface or with API commands):

- Telnet
- HTTP
- HTTPS
- SSH
- SNMP, may also be set to read only or traps only (only with API commands)

Disable Services using API Commands

Disable Telnet Service

`<xconfiguration Telnet Mode: Off>`

Disable HTTP Service

`<xconfiguration HTTP Mode: Off>`

Disable HTTP Service

`<xconfiguration HTTPS Mode: Off>`

Disable SSH Service

`<xconfiguration SSH Mode: Off>`

Disable SNMP Service

`<xconfiguration SNMP Mode: Off/ReadOnly/TrapsOnly>`

Disable Services using the Web Interface

1. To disable HTTP/HTTPS/Telnet/SSH, navigate to System Configuration > Misc > Services and set HTTPS, HTTPS, Telnet or SSH to Off.
2. Press Save button to save the changes.
3. Press Restart button. The service(s) will be deactivated at next restart.

Secure Shell (SSH)

The TANDBERG MPS supports SSH (Secure Shell) for secure access to the system.

SSH commands are encrypted and secured in several ways. Both ends of the client/server connection are authenticated using a digital certificate, and passwords are protected by being encrypted.

Read about the API commands in the TANDBERG MPS API Guide. Go to: http://www.tandberg.com/docs.

Security Alert

The system can notify management applications when someone tries remote access over IP with illegal password (via SNMP traps).

Information about the intruder's IP-address and the service used (Web, Telnet and FTP) will be given.

When the (optional) TANDBERG Management Suite (TMS) is used, an email notification may also be sent e.g. to the network administrator.

Product Approvals and CE Declarations

The product has been approved by various international approval agencies, among others: CSA and Nemko. According to their Follow-Up Inspection Scheme, these agencies also perform production inspections at a regular basis, for all production of TANDBERG's equipment.

The test reports and certificates issued by the approval agencies show that the TANDBERG MPS complies with the following standards.

EMC Emission - Radiated Electromagnetic Interference

- EN55022:1994 + A1:1995 + A2:1997 (CISPR 22:1993 + Corr. and Am.1 and Am.2) Class A (Comply with EU's Commission Decision 89/336/EEC).
- FCC Rules and Regulations Part 15, Subpart B, Class A.

EMC Immunity

- EN 55024:1998 + A1:2001 + A2: 2003
- EN 61000-3-2:2000
- EN 61000-3-3:1995 + A1:2001
  - (Comply with EU's Commission Decision 89/336/EEC).

Electrical Safety

- IEC 60950-1 edition 2001
- EN 60950-1 edition 2001 +A11:2004
- UL 60950-1. 1st Edition
- CSA 60950-1-03

EC DECLARATION OF CONFORMITY TANDBERG MPS 200

MANUFACTURER: TANDBERG Telecom AS
PRODUCT NAME: TANDBERG MPS 200
TYPE NUMBER: TTC3-02
DESCRIPTION: Media Processing System

This product complies with Commission Directives:
LVD 73/23/EEC
EMC 89/336/EEC
R&TTE 99/5/EEC

This product complies with harmonized standards:
EN 60950-1 : 2001
EN 55022 : 1994, A1/A2
EN 55024 : 1998, A1/A2
EN 61000-3-2 : 2000
EN 61000-3-3 : 1995, A1
TBR 4 Layer 1, 2 and 3

TECHNICAL CONSTRUCTION FILE NO.: X13513
YEAR WHICH THE CE-MARK WAS AFFIXED: 2005

For an official, signed version of this document, or details regarding documentation from the technical construction file, please contact TANDBERG

EC DECLARATION OF CONFORMITY TANDBERG MPS 800

MANUFACTURER: TANDBERG Telecom AS
PRODUCT NAME: TANDBERG MPS 800
TYPE NUMBER: TTC3-01
DESCRIPTION: Media Processing System

This product complies with Commission Directives:
LVD 73/23/EEC
EMC 89/336/EEC
R&TTE 99/5/EEC

This product complies with harmonized standards:
EN 60950-1 : 2001
EN 55022 : 1994, A1/A2
EN 55024 : 1998, A1/A2
EN 61000-3-2 : 2000
EN 61000-3-3 : 1995, A1
TBR 4 Layer 1, 2 and 3

TECHNICAL CONSTRUCTION FILE NO.: D13391
YEAR WHICH THE CE-MARK WAS AFFIXED: 2004

For an official, signed version of this document, or details regarding documentation from the technical construction file, please contact TANDBERG.

A Class Product Declaration

声　明

此为 A 级产品，在生活环境中，该产品可能会造成无线电干扰。在这种情况下，可能需要用户对其干扰采取切实可行的措施。

WARNING:
This is a class A product. In a domestic environment this product may cause radio interference in which case the user may be required to take adequate measures.

SIP - Current RFC's and Drafts Supported

The following current RFC's and drafts are supported:

- RFC 1889 RTP: A Transport Protocol for Real-time Applications
- RFC 2190 RTP Payload Format for H.263 Video Streams
- RFC 2327 SDP: Session Description Protocol
- RFC 2396 Uniform Resource Identifiers (URI): Generic Syntax
- RFC 2429 RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)
- RFC 2617 Digest Authentication
- RFC 2782 DNS RR for specifying the location of services (DNS SRV)
- RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 2976 The SIP INFO Method
- RFC 3016 RTP Payload Format for MPEG-4 Audio/Visual Streams
- RFC 3047 RTP Payload Format for ITU-T Recommendation G.722.1
- RFC 3261 SIP: Session Initiation Protocol
- RFC 3262 Reliability of Provisional Responses in SIP
- RFC 3263 Locating SIP Servers
- RFC 3264 An Offer/Answer Model with SDP
- RFC 3311 UPDATE method
- RFC 3361 DHCP Option for SIP Servers
- RFC 3420 Internet Media Type message/sipfrag
- RFC 3515 Refer method
- RFC 3550 RTP: A Transport Protocol for Real-Time Applications
- RFC 3581 Symmetric Response Routing
- RFC 3605 RTCP attribute in SDP
- RFC 3711 The Secure Real-time Transport Protocol (SRTP)
- RFC 3840 Indicating User Agent Capabilities in SIP
- RFC 3890 A Transport Independent Bandwidth Modifier for SDP
- RFC 3891 The SIP "Replaces" Header
- RFC 3892 Referred-By Mechanism
- RFC 3960 Early Media
- RFC 3984 RTP Payload Format for H.264 Video
- RFC 4028 Session Timers in SIP
- RFC 4145 TCP-Based Media Transport in the SDP
- RFC 4566 SDP: Session Description Protocol
- RFC 4568 SDP:Security Descriptions for Media Streams
- FC 4574 The Session Description Protocol (SDP) Label Attribute
- RFC 4582 The Binary Floor Control Protocol
- RFC 4583 Format for Binary Floor Control Protocol (BFCP) Streams
- RFC 4585 Extended RTP Profile for RTCP-Based Feedback
- RFC 4587 RTP Payload Format for H.261 Video Streams
- RFC 4629 RTP Payload Format for ITU-T Rec. H.263 Video
- RFC 4796 The Session Description Protocol (SDP) Content Attribute
- draft-ietf-xcon-bfcp-connection-04.txt
- draft-levin-mmusic-xml-media-control-08.txt
- draft-ietf-sipping-cc-transfer-07.txt
- draft-kristensen-avt-rtp-h264-extension-00.txt

Configuring LCS and MPS for SIP

Configuring Microsoft Office Live Communications Server (LCS) 2005, SP1

This section will illustrate how to setup the Live Communication Server (LCS) to work properly with the MPS.

SIP (Session Initiated Protocol)

Software version J3 and above for the TANDBERG MPS has support for the Session Initiated Protocol (SIP).

The first version of support will be compatible with the Microsoft Live Communications Server (LCS) 2005, SP1.

LCS will serve as both an authentication authority and a SIP Proxy to the MPS for both incoming and outgoing SIP calls.

Configuring LCS and MPS

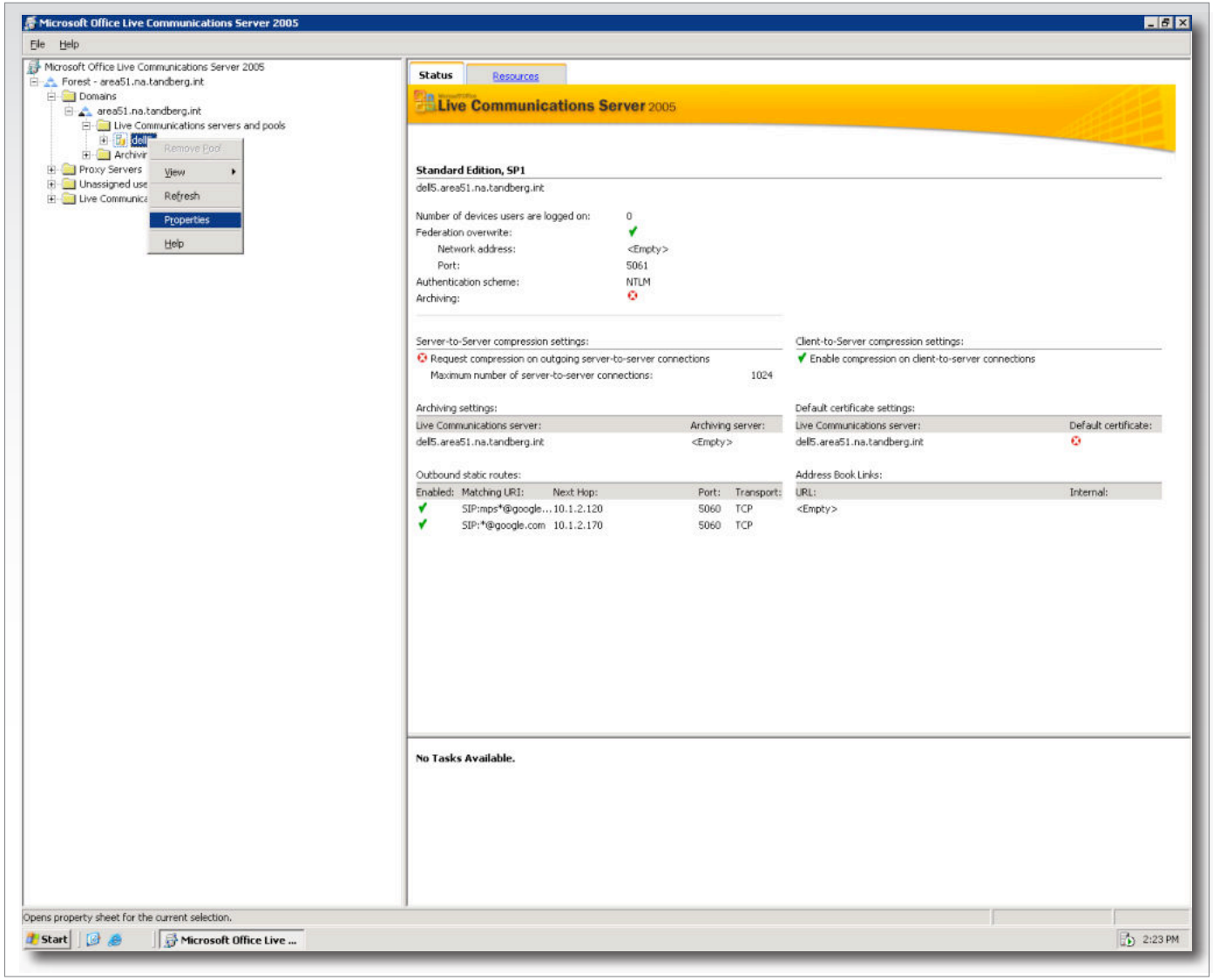
1. The Microsoft Live Communication Server (LCS) should be configured prior to configuring the MPS unit.
2. In order to facilitate proper communication between the MPS and LCS, the LCS must be configured to see the MPS as a trusted site as well as setup the dial-plan to forward appropriate calls to the MPS.

Configuring LCS

In order for the MPS to receive calls on the SIP network, the LCS is configured to allow the MPS to receive multiple calls on multiple Dial In numbers.

LCS dialling works on a prefix basis:

- A specific prefix is assigned to forward all calls to a specific host. This prefix can be anything and is usually followed by a wildcard identifier *.



Configuring LCS and MPS for SIP, *cont...*

Configuring Microsoft Office Live Communications Server (LCS) 2005, SP1

This section will illustrate how to setup the Live Communication Server (LCS) to work properly with the MPS.

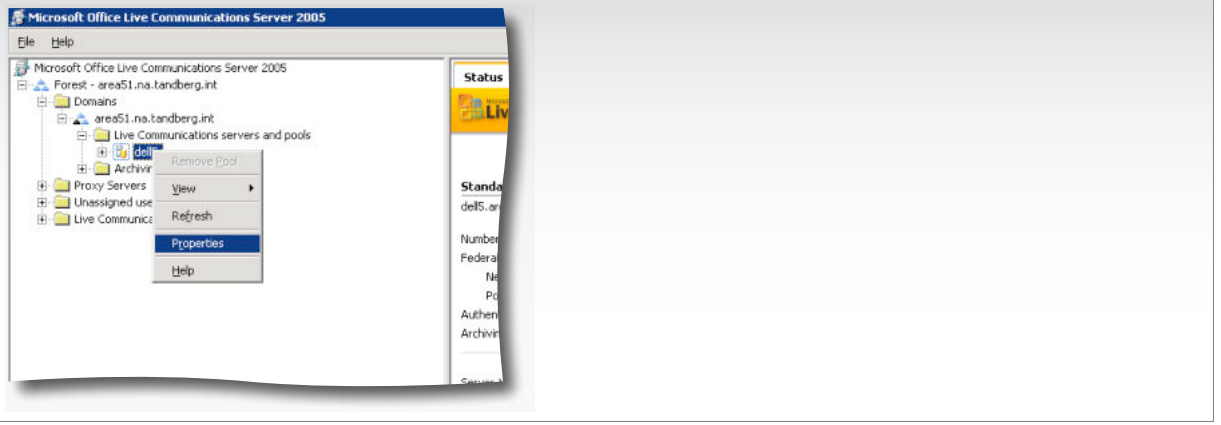
Login and Locate the Server

1. Login to the Live Communication Server
2. Open the Live Communication Server snap-in for MMC.
3. Expand the tree to locate the server you wish the MPS to use as its home server.
4. Once located right-click on the server's icon and select Properties.
5. In the Properties dialog select the Host Authorization tab to add a new authorized (trusted) host.

Add a Trusted Host

This setup will allow the LCS to authenticate the MPS and treat it as a trusted host within LCS

1. Select the Host Authorization tab
2. Press the Add... button to add a new authorized host.
3. In the Add Authorized Host window, set the server selection to IP Address and enter the IP address of the System Controller Board of the MPS.
4. The selection boxes for Throttle As Server and Treat As Authenticated should be checked.
5. Once the trusted host configuration has been completed, press OK to save the changes.



Configuring LCS and MPS for SIP, *cont...*

Configuring Microsoft Office Live Communications Server (LCS) 2005, SP1

This section will illustrate how to setup the Live Communication Server (LCS) to work properly with the MPS.

Add a Route

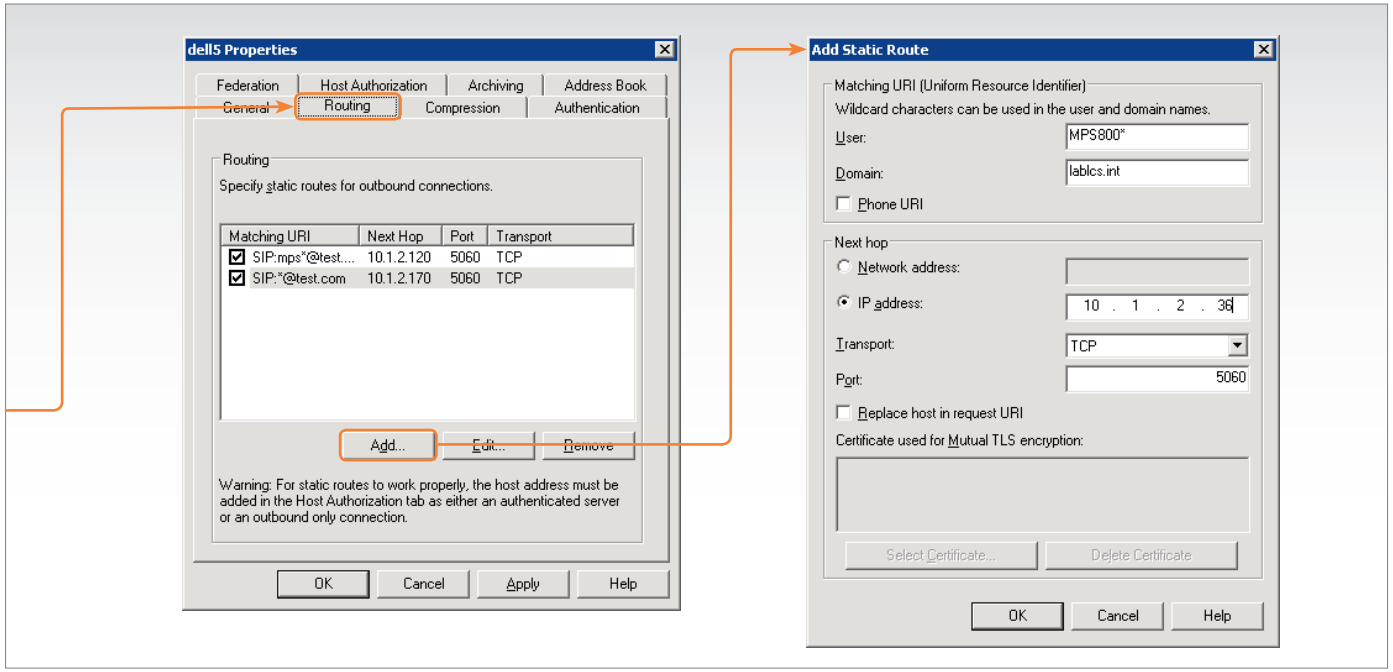
When the MPS is configured as an Authorized Host to the LCS, the LCS needs to be told what SIP URIs should be forwarded to the MPS. This is done by configuring a route in the Live Communication Server.

The SIP URI to reach the MPS will be the combination of the User and Domain fields.

Under the same server properties window (see previous page), select the Routing tab and press Add... to open the Add Static Route dialog and add a new routing pattern:

1. The User field is used to specify which URIs will be forwarded by this rule. Wildcards (*) are allowed.
2. In the Domain box, enter the domain that will be used with the SIP URI for the MPS. The SIP URI to reach the MPS will be the combination of the user and domain fields. Example: <MPS@company.com>
3. Under Next Hop, select IP Address and enter the IP address of the System Controller Board of the MPS.
4. Set Transport to TCP
5. Set Port to 5060.
6. Press the OK button to save the route
7. Press the OK button to close the properties window of the server.

This completes the configuration of the LCS server. Next, the MPS will need to be configured to use the LCS server.



Example #1:

If you want all calls coming in to <MPS> to be forwarded to the MPS, enter <MPS> in the User field.

LCS does allow a wildcard entry. Enter the <*> character if a wildcard entry is desired. For example, if all calls beginning with <MPS> should be forwarded to the MPS itself (e.g. <MPS123@lcs.int>, <MPS456@lcs.int>, <MPSConference2@lcs.int>), then enter <MPS*> in the user field.

In the Domain box, enter the domain that will be used with the SIP URI for the MPS. The complete SIP URI to reach the MPS will be the combination of the user and domain fields. Example: <MPS@company.com>

Under Next Hop, select IP Address and enter the IP address of the controller card of the MPS. Make sure Transport is set to TCP and Port is 5060. The completed route should look like the figure to the left. Press OK to save the route, and press OK again to close the Properties window of the server.

Configuring LCS and MPS for SIP, *cont...*

SIP Configuring on the MPS

This section will illustrate how to setup the MPS for SIP. The Microsoft Live Communication Server (LCS) should be configured prior to the MPS unit.

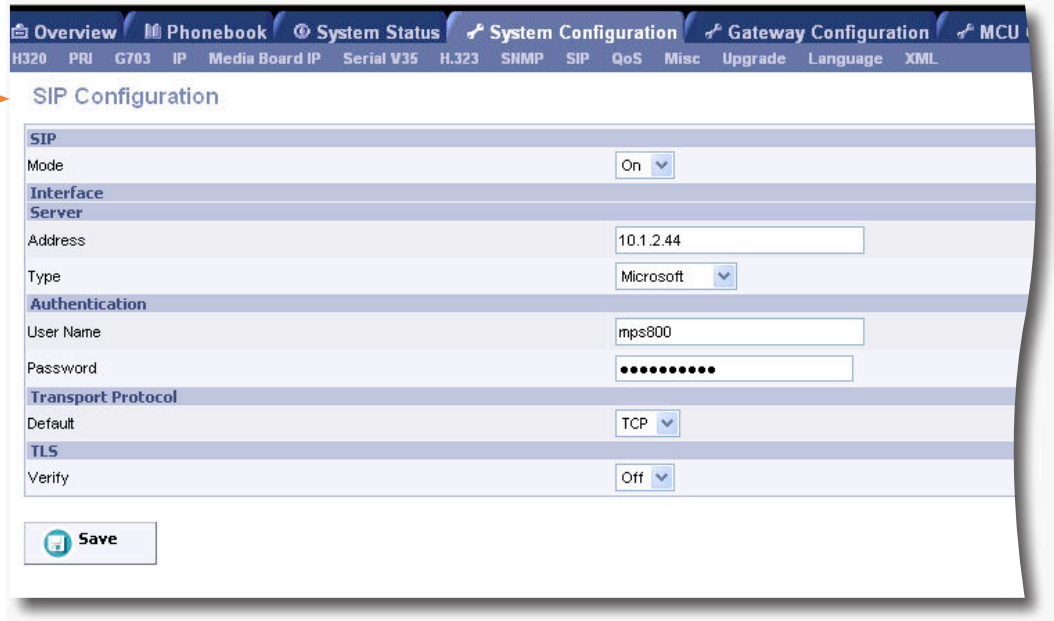
The MPS configuration consists of ensuring the MPS has a valid DNS name server configured and entering the LCS server information.

SIP Configuration

To configure the MPS for SIP communication, navigate to System Configuration > SIP:

1. Set SIP Mode to On
2. Configure the Interface for SIP communication. Enter the DNS Name or IP Address for the SIP server in the Address field and set the SIP Server Type to Microsoft
3. Press the Save button for the changes to take effect.

Please refer to section System Configuration > SIP Configuration for more details.



Configuring LCS and MPS for SIP, *cont...*

DNS Server Configuration on the MPS

The LCS SIP operation requires the MPS to have a valid DNS (Domain Name System) server configured.

If the DNS server is not configured properly, calls may connect to the MPS, but neither video nor audio will be transmitted to the far end.

DNS Configuration

To configure the DNS server on the MPS, navigate to System Configuration > IP:

Locate the configuration for the particular Service Controller Interface the system will use and enter the IP Address of the Domain Name Server (DNS) Interface 1. Up to five DNS addresses can be configured.

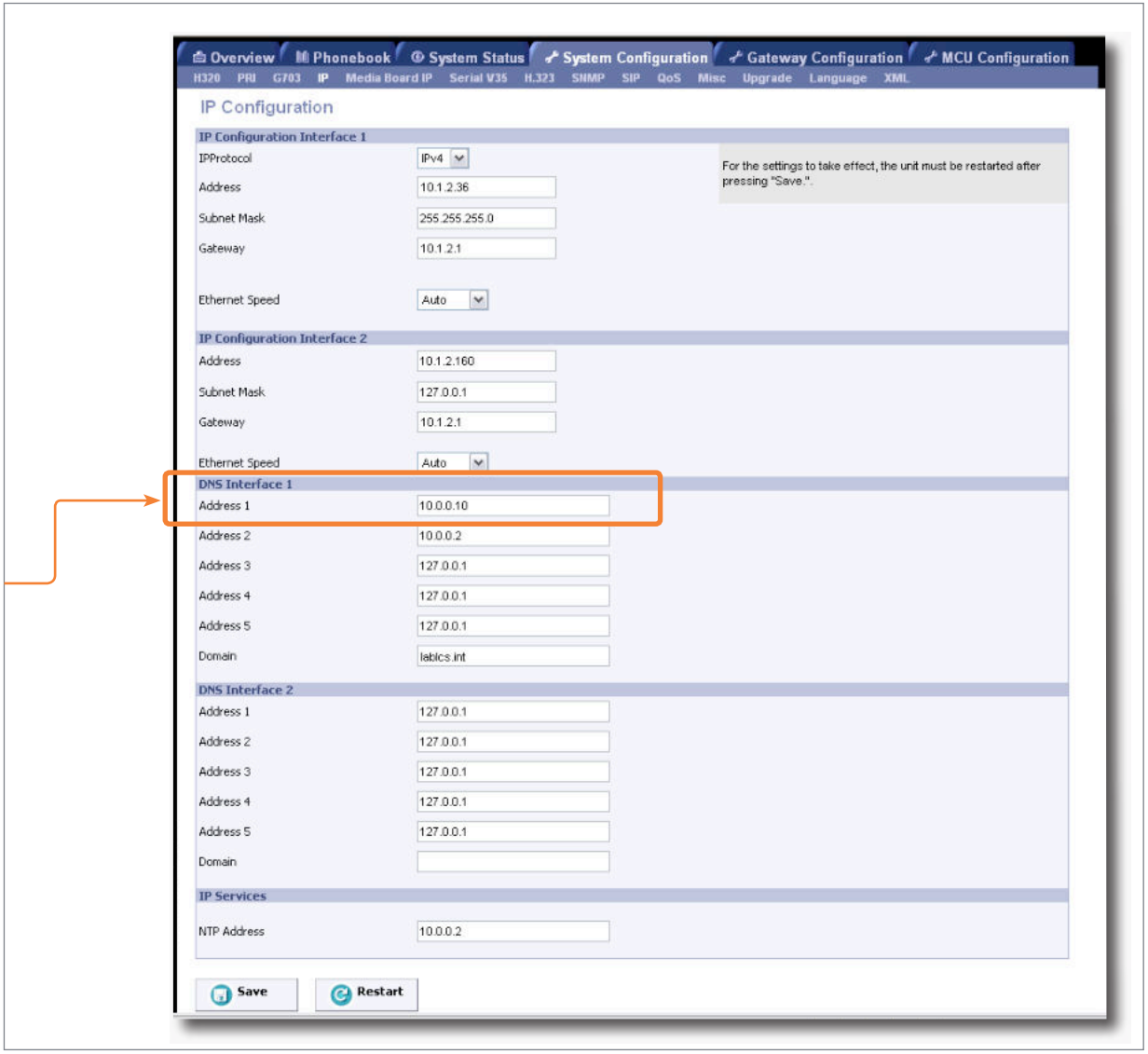
Reboot the MPS

After having configured both the SIP and IP settings you must reboot the MPS.

- Press the Restart button to reboot the MPS.

SIP is only supported on DNS Interface 1.

Please refer to section System Configuration > IP Configuration for more details.



Configuring LCS and MPS for SIP, *cont...*

Verify the Configuration

Once the MPS has restarted, the Overview page of the MPS should show that the SIP status is Server Active.

Verify that the configuration is active before you continue.

Configuring SIP Dial In Numbers

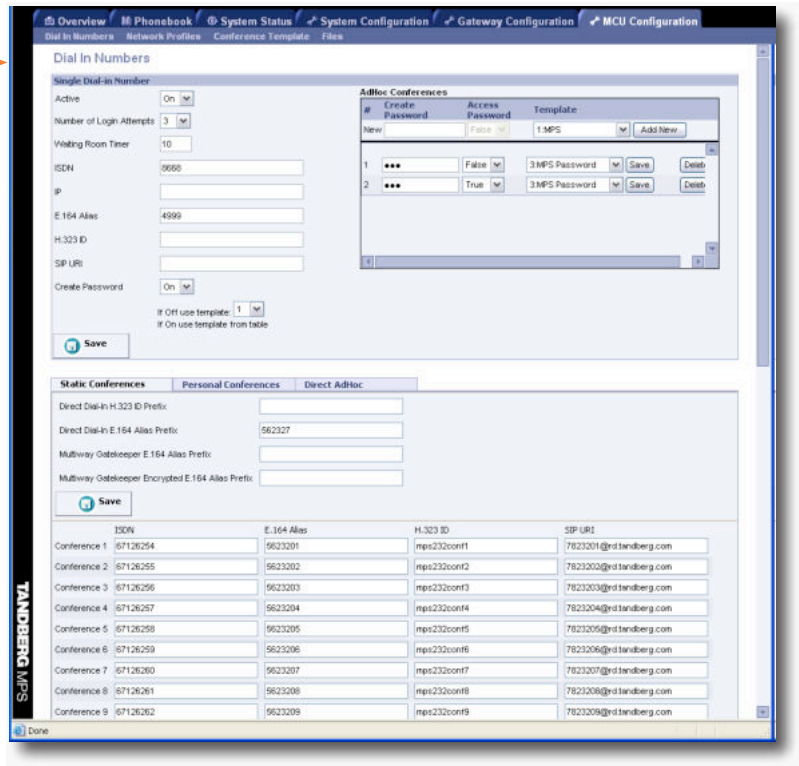
When Server Active has been verified, it is time to configure the appropriate SIP Dial In Numbers for the MPS system. Navigate to MCU Configuration > Dial In Numbers.

Within this screen, enter all the appropriate SIP URIs for the appropriate Dial In Conferences and Single Dial In Numbers.

All SIP URIs, configured in the MPS, must be fully qualified domain names with the domain name that is configured under the Routing configuration within the LCS.

The prefix, entered under the Routing configuration, must be included within all of the SIP names for the MPS to properly receive the calls.

Once all of the dial-in URIs are configured, the MPS is ready to make and receive SIP calls from other SIP devices registered to the LCS network.



Example #1:

If the domain name for the LCS server is <lcs.int> and the SIP prefix for the MPS was configured as <MPS*>, your dial in numbers must begin with <MPS> and end with <@lcs.int> in order to be valid, e.g.:

<MPSConference1@lcs.int>

<MPSConference2@lcs.int>

<MPSDialin@lcs.int>

<MPS12345@lcs.int>

Configuring OCS 2007 and MPS for SIP

Configuring Microsoft Office Communications Server (OCS) 2007

This section will illustrate how to setup the Office Communication Server (OCS) to work properly with the MPS

SIP (Session Initiated Protocol)

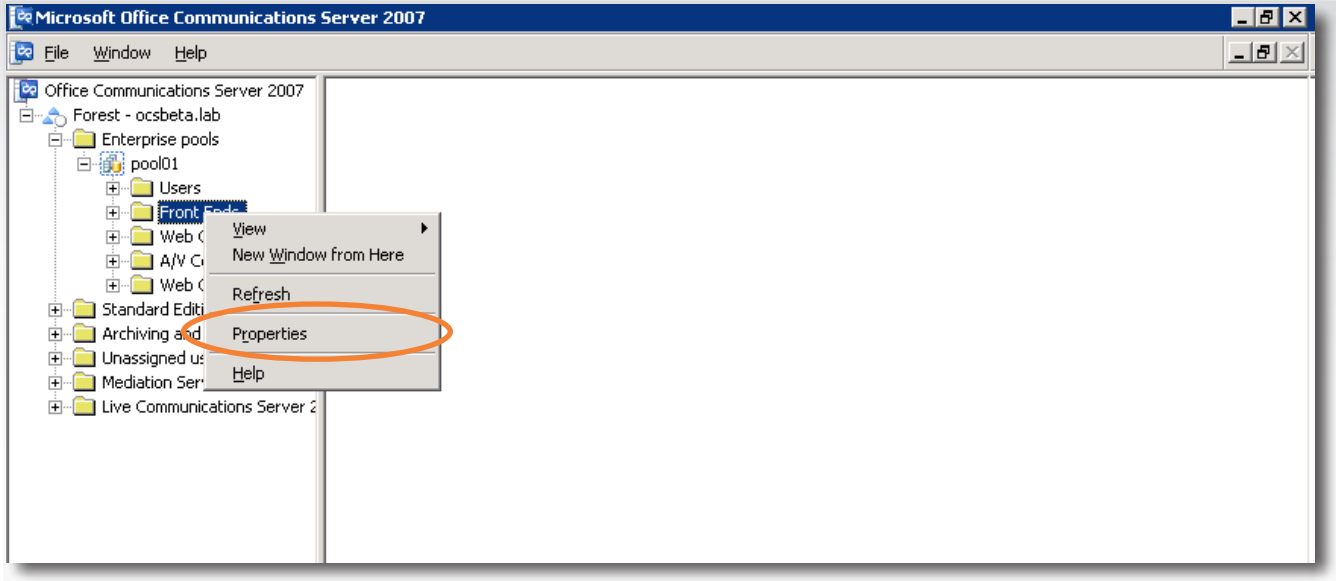
Software version J3 and above for the TANDBERG MPS has support for the Session Initiated Protocol (SIP).

Todays version of support is compatible with the Microsoft Office Communications Server (OCS) 2007.

OCS will serve as both an authentication authority and a SIP Proxy to the MPS for both incoming and outgoing SIP calls.

Configuring OCS and MPS

1. The Microsoft Office Communication Server (OCS) should be configured prior to configuring the MPS unit.
2. In order to facilitate proper communication between the MPS and OCS, the OCS must be configured to see the MPS as a trusted site as well as setup the dial-plan to forward appropriate calls to the MPS.



TCP Connections

TCP connections are not enabled on OCS by default. If not done already, you need to open for TCP connections.

Configuring the TCP connections:

1. Expand the Front Ends and right-click the OCS Server which you want to configure.
2. Click Properties for the chosen server and then click Add... to configure the TCP connections:
   - IP Address: All
   - Port: 5060
   - Transport: TCP

Configuring OCS and MPS for SIP, *cont...*

Configuring Microsoft Office Communications Server (OCS) 2007

This section will illustrate how to setup the Office Communication Server (OCS) to work properly with the MPS

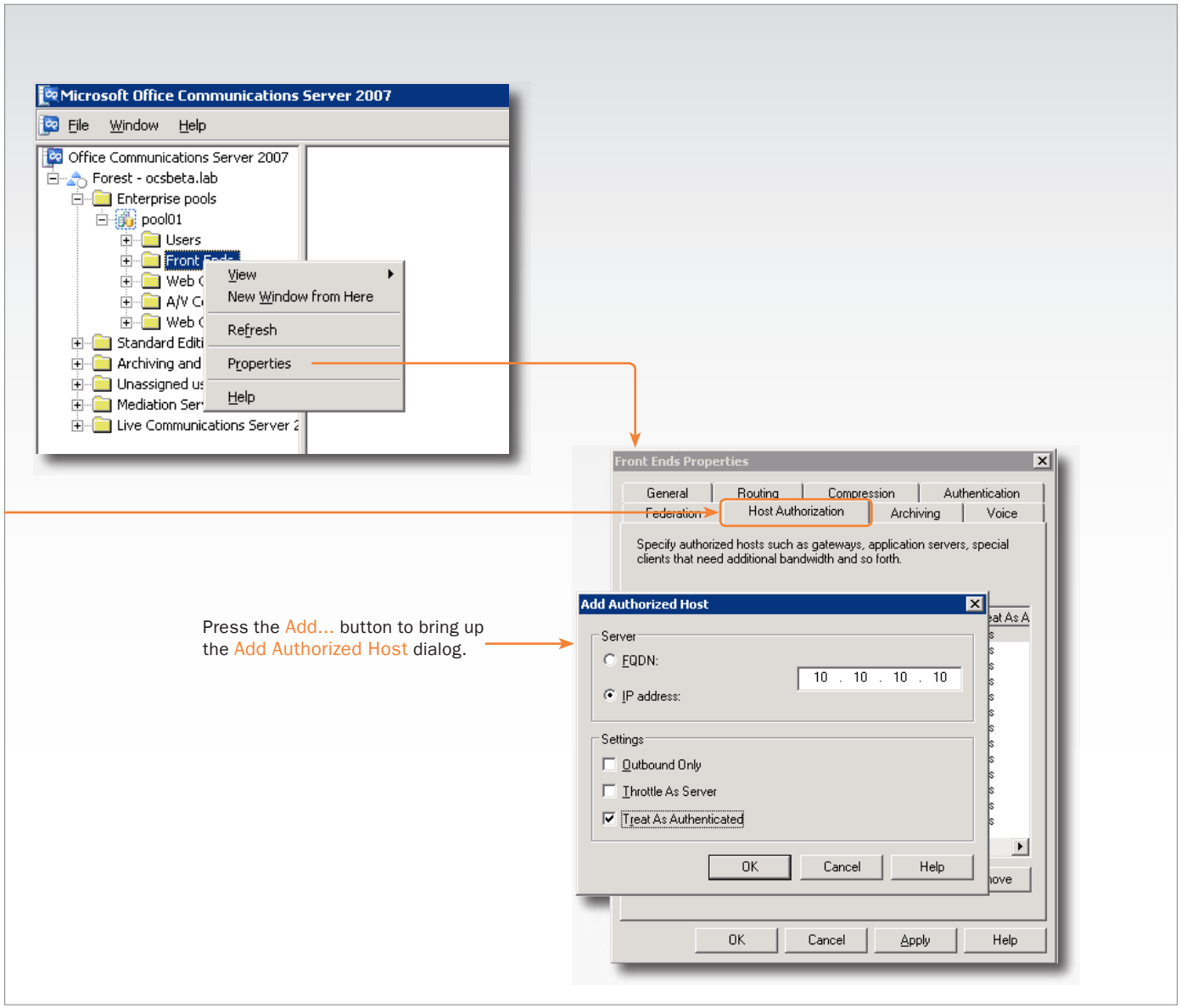
Login and Locate the Server

1. Log on to the Office Communications Server 2007 server as a member of the RTCUniversalServerAdmins group.
2. Open Office Communications Server 2007.
3. In the console tree, expand the forest node, and then do one of the following:
   - For an Enterprise pool, expand Enterprise pools, expand the pool, right-click Front Ends, and then click Properties.
   - For a Standard Edition Server, expand Standard Edition servers, right-click the pool, click Properties, and then click Front End Properties.

Add a Trusted Host

This setup will allow the OCS to authenticate the MPS and treat it as a trusted host within OCS

1. In the Front End Properties select the Host Authorization tab, press the Add... button to add a new authorized host.
   a. In the Add Authorized Host window, set the server selection to IP Address and enter the IP address of the System Controller Board of the MPS.
   b. The selection box Treat As Authenticated must be checked.
2. Once the trusted host configuration has been completed, press OK to save the changes.



Configuring OCS and MPS for SIP, *cont...*

Configuring Microsoft Office Communications Server (OCS) 2007

This section will illustrate how to setup the Office Communication Server (OCS) to work properly with the MPS

Add a Route

When the MPS is configured as an Authorized Host to the OCS, the OCS needs to be told what SIP URIs should be forwarded to the MPS. This is done by configuring a route in the Office Communication Server.

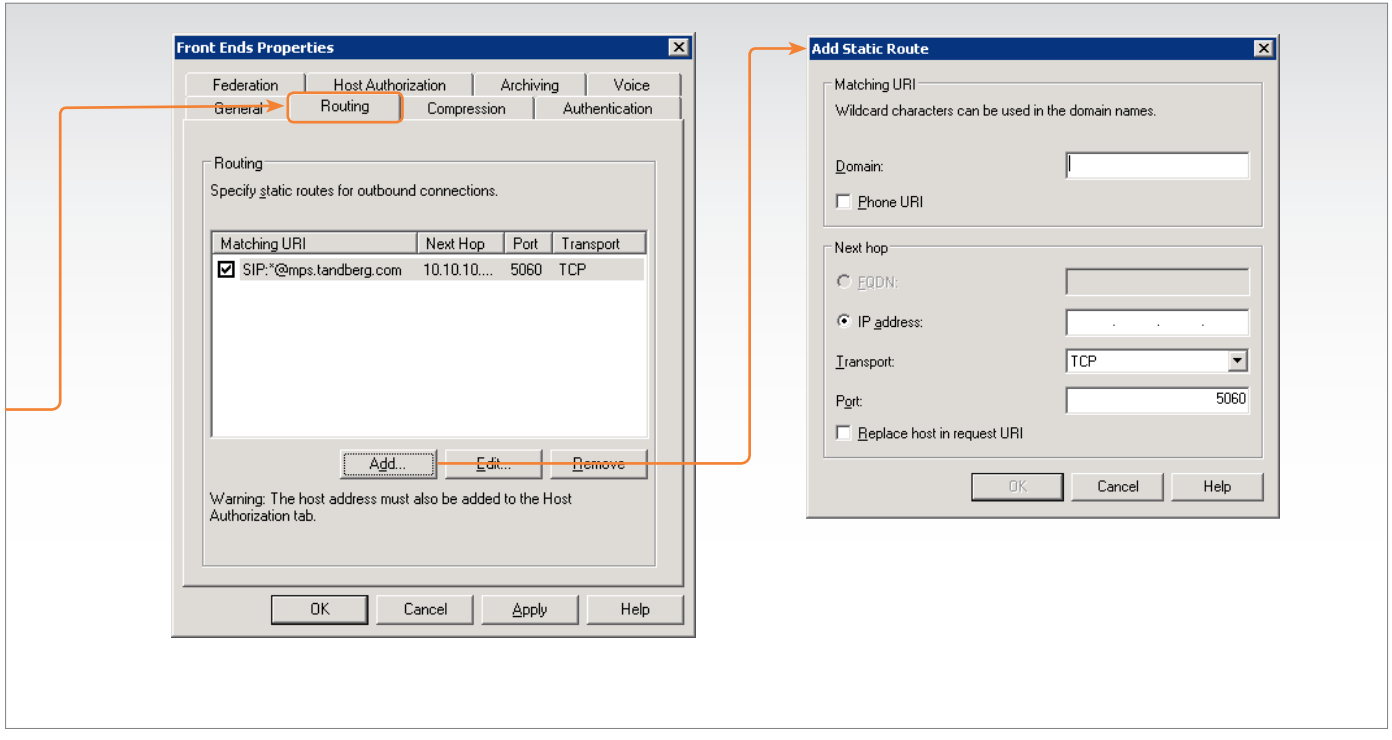
The SIP URI to reach the MPS will be the combination of the User and Domain fields.

Under the same server properties window (see previous page), select the Routing tab and press Add... to open the Add Static Route dialog and add a new routing pattern:

1. In the Domain box, enter the domain that will be used with the SIP URI for the MPS. The domain name must be a unique domain name and **must correspond** with the domain name given in the MCU Configuration. The domain name is often maid unique by adding a prefix. Example: `mps.company.com`
2. Under Next Hop, select IP Address and enter the IP address of the System Controller Board of the MPS.
3. Set Transport to TCP
4. Set Port to 5060.
5. Press the OK button to save the route
6. Press the OK button to close the properties window of the server.

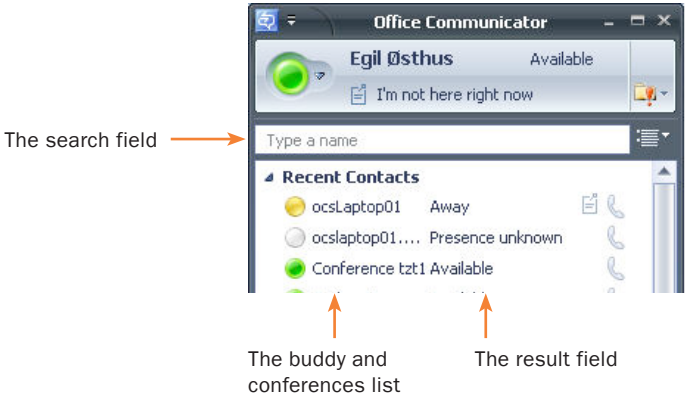
Unless you need to open for TCP connections, this completes the configuration of the OCS server.

Wait for some time for the new routes and host authorization lists to be updated.



Using the Microsoft Office Communicator Client (MOC)

1. Enter the name of the personal conference in the Search field, just below the contact information field in your MOC client.
2. The presence information should be propagated from the MPS, as Available if the conference is available.
3. Right-click on the buddy that appears in the results field, and click Start a Video Call.
4. To add the virtual meeting to your buddy list, right click on the buddy that appears in the results field and choose Add to contact list.



Configuring OCS and MPS for SIP, *cont...*

Configuring the MPS

The Microsoft Office Communication Server (OCS) should be configured prior to the MPS unit.

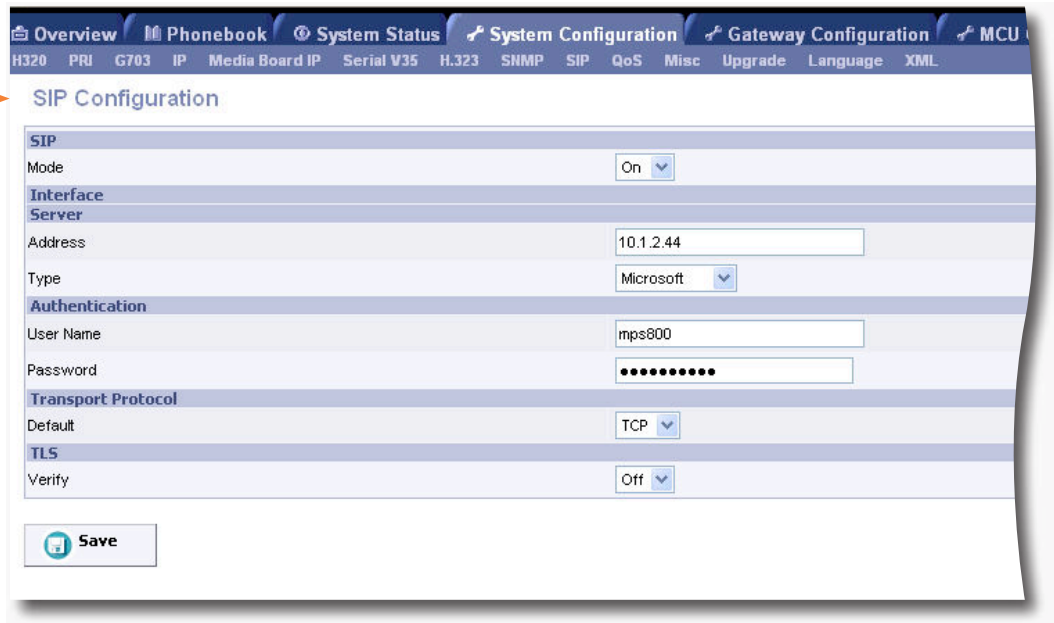
The MPS configuration consists of ensuring the MPS has a valid DNS Server configured and entering the Office Communication Server (OCS) information.

SIP Configuration

To configure the MPS for SIP communication, navigate to System Configuration > SIP:

1. Set SIP Mode to On
2. Set the Address to the Office Communications Server (OCS). This address must be a fully qualified domain name or an IP address.
3. Set the Type to Microsoft
4. Set the Transport Protocol to TCP (default)
5. Set the TLS Verify to Off
6. Press the Save button for the changes to take effect.

Please refer to section System Configuration > SIP Configuration for more details.



Configuring OCS and MPS for SIP, *cont...*

DNS Server Configuration

The OCS SIP operation requires the MPS to have a valid DNS (Domain Name System) server configured.

If the DNS server is not configured properly, calls may connect to the MPS, but neither video nor audio will be transmitted to the far end.

DNS Configuration

To configure the DNS server on the MPS, navigate to System Configuration > IP:

Locate the configuration for the particular Service Controller Interface the system will use and enter the IP Address of the Domain Name Server (DNS) Interface 1. Up to five DNS addresses can be configured.

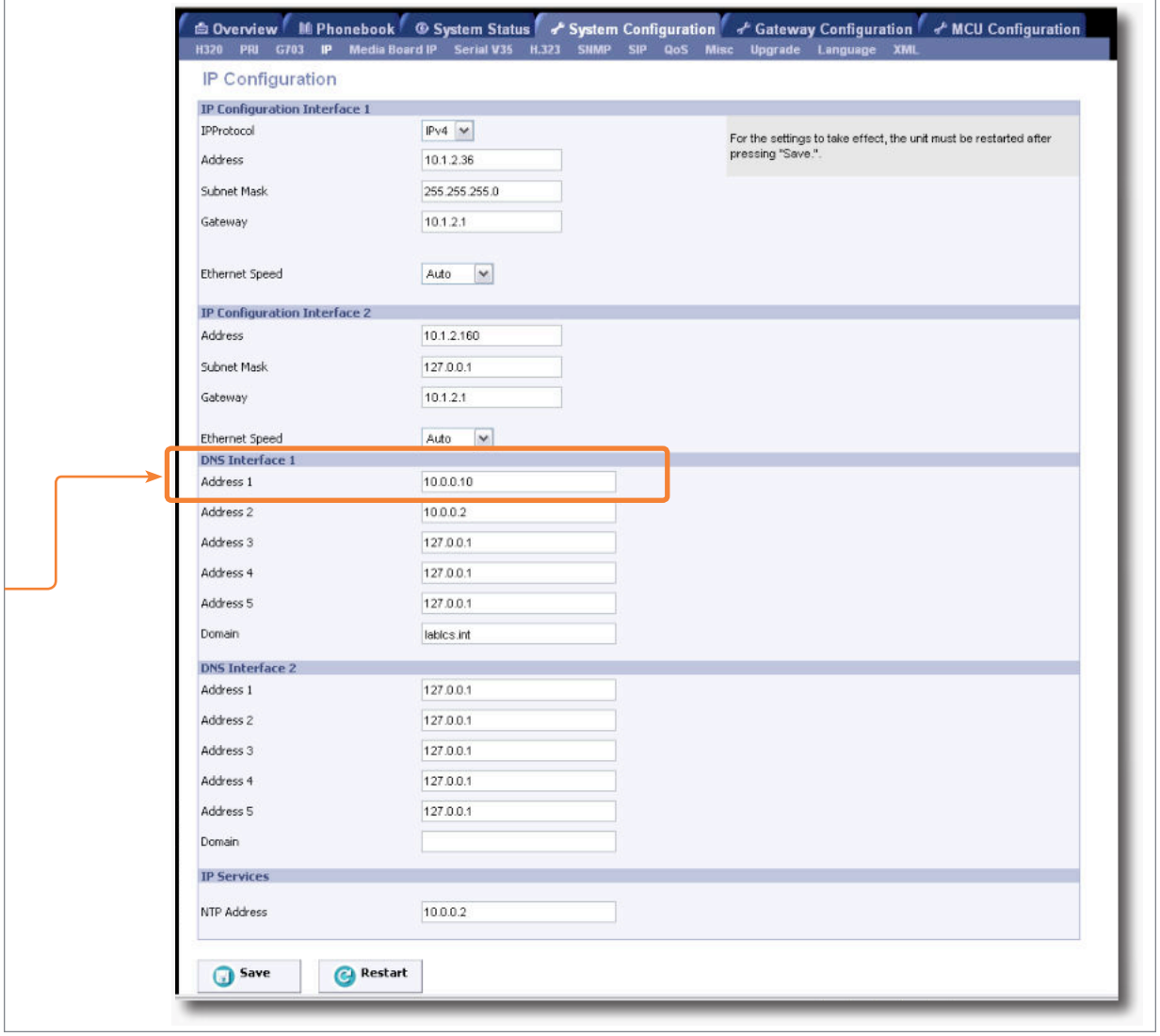
Reboot the MPS

After having configured both the SIP and IP settings you must reboot the MPS.

- Press the Restart button to reboot the MPS.

SIP is only supported on DNS Interface 1.

Please refer to section System Configuration > IP Configuration for more details.



Configuring OCS and MPS for SIP, *cont...*

Verify the Configuration

Once the MPS has restarted, the Overview page of the MPS should show that the SIP status is Server Active.

Please verify that the configuration is active before you continue.

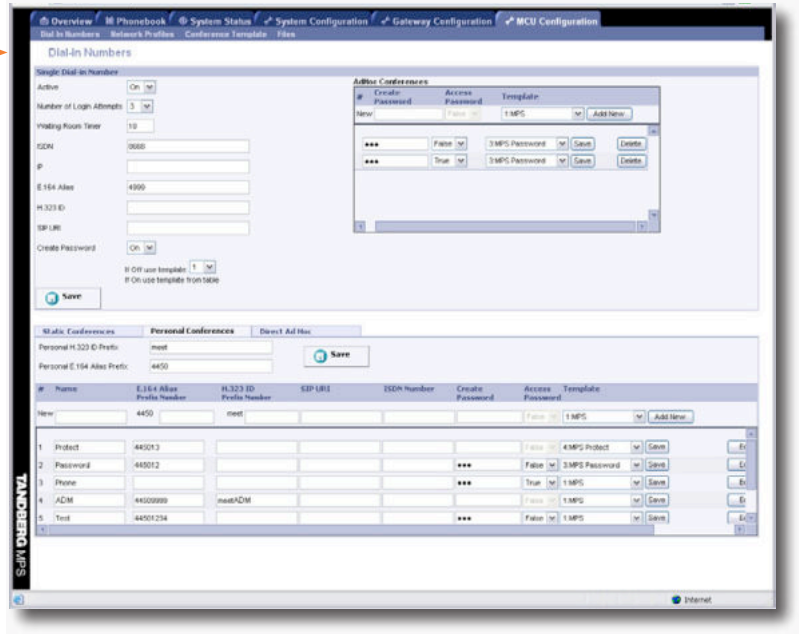
Configuring SIP Dial In Numbers

When Server Active has been verified, it is time to configure the appropriate SIP Dial In Numbers for the MPS system. Navigate to MCU Configuration > Dial In Numbers > Personal Conferences.

Within this screen, enter all the appropriate SIP URIs.

All SIP URIs, configured in the MPS, must be fully qualified domain names with the domain name that is configured under the Routing configuration within the OCS.

Once all of the dial-in URIs are configured, the MPS is ready to make and receive SIP calls from other SIP devices registered to the OCS network.



Example #1:

If the domain name of the OCS server is <ocs.int>, and your sub-domain for MPS calls is <mps.ocs.int>, your dial in numbers use the domain <mps.ocs.int>, e.g.:

"Conference1@mps.ocs.int" Conference1@mps.ocs.int

Technical Specifications

TANDBERG MPS 200

MAXIMUM SYSTEM CAPACITY
MCU video ports: up to 40
MCU audio ports: up to 32
Gateway sessions: up to 20
Serial Interface ports: up to 64
ISDN-PRI ports: 16

MCU CONFERENCE SCHEDULING
Advanced scheduling (Multiple MCUs) with TMS
Scheduling using the TANDBERG Scheduler, Microsoft Outlook® or IBM Lotus Notes®
On-demand conferencing (true Ad Hoc)
Personal conferencing
Conference initiation using Microsoft Office Communicator (SIP-CX)

MCU CONFERENCE MANAGEMENT
Advanced conference monitoring with TMS Conference Control Center
Centralized management and control via embedded web server
3rd Party API for external control (XML on HTTPS)
Live audio and video monitoring & diagnostics
Dual Snapshot of ongoing conference (JPEG)
Move participants between conferences
Auto dial out on personal conferences
Waiting room provided to attendees prior to conference start
Conference bandwidth threshold
Single or Multiple Dial In numbers per conference
10 Prefixes for Ad Hoc Conferencing
Caller ID conference routing
Mute audio and video to/from endpoints
Custom “Welcome” pages (JPEG)
Customizable conference messages
Lock participant layouts
Edit conference display text
Access to all conferences through Single Number Dial In Interface (SNDI)

MCU VIDEO FEATURES
Up to 30 fps in Continuous Presence, Voice Switched and presentation mode
Best Impression — Automatic change of video image layout and resolution
Send/recieve wide format
Automatic Continuous Presence switching
Common Layouts for 16:9 and 4:3 Formats
Standard CP (CP2, CP3, CP4, CP5+1, CP4+3, CP7+1, CP9, CP8+2, CP12+1, CP12+2, CP16/auto)
Wide Screen CP (wCP2+1,wCP3, wCP3+1, wCP4, wCP4+1, wCP6, wCP8+1, wCP12, CP32)
Voice Switched
Self-view optional
Lecture mode
Embedded site naming with Unicode support
Active speaker indication in CP mode
Active speaker indication for telephones

MCU CHAIR CONTROL
Embedded control from TMS Conference Control Center and embedded web server
Endpoint control using H.243 Chair Control on H.320 and H.323
Endpoint control using BFCP Chair Control on SIP
Multi-language H.243 terminal names

MCU AUDIO FEATURES
Mute Audio to/from endpoints
Automatic gain control
Audio input level indications
Custom “welcome” sound (WAV)
Customizable conference messages (WAV)
Entry/Exit tones
Mobile telephone noise suppression
Telephone echo suppression
CD-quality audio using MPEG 4 AAC-LD

GATEWAY FEATURES (OPTION)
H.323 to/from H.320 (IP to ISDN)
H.323 to/from H.320 (IP to V.35)
GATEWAY ISDN CALL ROUTING
Direct Inward Dialling (DID)
Interactive Voice and Video Response (IV2R)
Automatic TCS-4 response on ISDN for automated bridging of IP islands
Any combination of DID, IV2R and TCS-4
Automatic downspeeding to available ISDN or IP rate

GATEWAY H.323 CALL ROUTING TO ISDN
Single prefix dialling
Selectable 2x64k (H.221) dial-out
Automatic ISDN restrict
Automatic downspeed to available ISDN or IP rate
Load balance control
Call transfer of the IP side

ENDPOINTS SUPPORTED
ISDN (H.320) video endpoints up to 2 Mbps
V.35 (H.320) video endpoints up to 2 Mbps
IP (H.323) video endpoints up to 2 Mbps
IP (SIP) video endpoints up to 2 Mbps
IP (3G) video phone up to 64 kbps
Microsoft Office Communicator (SIP-CX)*
Analog/IP telephony

EMBEDDED ENCRYPTION
Standards based on ISDN, IP and mixed ISDN/IP: H.233, H.234, H.235 v2&v3, DES 56 bit key, AES 128 bit key
NIST-validated AES and DES
Automatic key generation and exchange
Mix of DES/AES possible in the same conference

LIVE VIDEO RESOLUTIONS
NATIVE PAL:
QCIF (176 x 144 pixels)
CIF (352 x 288 pixels)
448p (576 x 448 pixels)*
4CIF (704 x 576 pixels)
NATIVE NTSC:
SIF (352 x 240 pixels)
400p (528 x 400 pixels)*
4SIF (704 x 480 pixels)
NATIVE PC RESOLUTIONS:
VGA (640 x 480 pixels)
SVGA (800 x 600 pixels)
XGA (1024 x 768 pixels)
NATIVE WIDESCREEN RESOLUTIONS:
w288p (512 x 288 pixels)
w448p (768 x 448 pixels)
w576p (1023 x 576 pixels)
w720p (1280 x 720p)

TRANSCODING
Optimal Voice-Switched (oVS) Video: Intelligent On-Demand Transcoding
Network transcoding (IP, ISDN-PRI and V.35)
Audio, video and protocol transcoding
Optimized picture layout for 3G phones
Bandwidths from 56 kbps — 2 Mbps in the same conference

SECURITY FEATURES
Network authentication (H.235)
Embedded audio/video encryption (H.235)
Secure management & control (HTTPS, SSH)
Participant authentication (password, Dial In number and caller ID)
IP administration passwords
Services may be disabled: Telnet, HTTP, HTTPS, SNMP, SSH

LIVE PRESENTATIONS
Broadcast of presentations (DuoVideo^{TF}, H.239 and BFCP**)

IP NETWORK FEATURES
Expressway NAT and firewall traversal
Intelligent Packet Loss Recovery (IPLR)
Automatic call rate adjustments
URI dialling
QoS (DiffServ, IP Precedence and TOS)
H.245 DTMF tones in H.323
Centralized management via embedded web server using HTTPS, XML, SOAP, SSH and Telnet
1 x RS-232 for local control and diagnostics
Endpoints on two different IP networks in the same conference
Automatic/manual error control
IPv6 Enabled - IPv4-IPv6 Dual Stack

ISDN NETWORK FEATURES
ISDN-PRI E1/T1
ISDN PRI G.703 and E1/T1, Leased Line
Non-Facilities Associated Signaling (NFAS)
Automatic downspeeding
Automatic call rate adjustments
Advanced diagnostic tools

Technical Specifications, *cont...*

TANDBERG MPS 200

Automatic/manual error control

NETWORK INTERFACES
2 x LAN/ethernet (RJ-45) 10/100 Mbit on the system controller board
Up to 2 x LAN/ethernet (RJ-45) 10/100 Mbit for media processing board
Up to 16 x E1/T1 (RJ-45) for ISDN PRIs and/or G.703
Up to 64 x V.35/RS-366 ports

ETHERNET/INTERNET/INTRANET CONNECTIVITY
TCP/IP, SSL, ARP, Telnet, HTTP, HTTPS, XML, SOAP
Embedded web server for total conference and call control
10/100 Mbps full/half duplex (manual or auto detect selection)

OTHER SUPPORTED STANDARDS
H.221, H.231, H.241, H.242, H.243, H.245, H.320, H.239, H.323, BONDING (ISO 13871), Q.931, SIP, H261, H263+, H264, G711, G.722, G.722.1, G728, MPEG4 AAC-LD

APPROVALS
Directive 73/23/EEC (Low Voltage Directive)
– Standard EN 60950
Directive 89/336/EEC (EMC Directive)
– Standard EN 55022, Class A
– Standard EN 55024
– Standard EN 61000-3-2/-3-3
Directive 1999/5/EEC (R&TTE Directive)
– Standard TBR4
Approved according to UL 60950 and CAN/CSA C22.2 No. 60950
Complies with FCC15B Class A

ENVIRONMENTAL DATA
Operating temperature:
0°C to 35°C (32°F to 95°F) ambient temperature
Relative Humidity (RH):
10% to 90%
Storage and transport temperature:
-20 to 60°C (-4°F to 140°F) at RH 10–90% (non-condensing)

POWER
100–240 VAC, 50/60 Hz;
460W max. power consumption

UNIT DIMENSIONS
Height 3U/133 mm/5.3”
Width 444 mm/17.4”
Depth 371 mm/14.6”
Net weight 20.9 lbs/9.5 kg
19” rack-mountable

* MCU only
** Binary Floor Control Protocol

Technical Specifications, *cont...*

TANDBERG MPS 800

MAXIMUM SYSTEM CAPACITY
MCU video ports: up to 160
MCU audio ports: up to 48
Gateway sessions: up to 80
Serial Interface ports: up to 128
ISDN-PRI ports: up to 32

MCU CONFERENCE SCHEDULING
Advanced scheduling (Multiple MCUs) with TMS Scheduling using the TANDBERG Scheduler, Microsoft Outlook® or IBM Lotus Notes®
On-demand conferencing (true Ad Hoc)
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TANDBERG MPS 800

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– Standard EN 60950
Directive 89/336/EEC (EMC Directive)
– Standard EN 55022, Class A
– Standard EN 55024
– Standard EN 61000-3-2/-3-3
Directive 1999/5/EEC (R&TTE Directive)
– Standard TBR4
Approved according to UL 60950 and CAN/CSA C22.2 No. 60950
Complies with FCC15B Class A

ENVIRONMENTAL DATA
Operating temperature:
0°C to 35°C (32°F to 95°F) ambient temperature
Relative Humidity (RH):
10% to 90%
Storage and transport temperature:
-20 to 60°C (-4°F to 140°F) at RH 10–90% (non-condensing)

POWER
100–240 VAC, 50/60 Hz;
800W max. power consumption

UNIT DIMENSIONS
Height 9U/399 mm/15.7”
Width 444 mm/17.4”
Depth 371 mm/14.6”
Net weight 44 lbs/20.0 kg
19” rack-mountable

* MCU only
** Binary Floor Control Protocol

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