

MX-ONE System Description

DESCRIPTION



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1 Introduction

Aastra MX-ONE is a complete communications solution. Not only does MX-ONE provide excellent voice and video communications, but it also provides the necessary applications to offer true Mobility and Unified Collaborative Communications. MX-ONE is based on an open software and hardware environment, using standard servers with a LINUX SUSE operating system.

Today's users are expected to have all types of communication services combined and integrated in the same application GUI. Furthermore users expect to have access to all services when they are on the move.

MX-ONE already set the standard for Fixed Mobile Convergence (FMC) with its mobile extension feature, where mobile phones are integrated as an integral part of the office communications system with access to a complete feature set.

MX-ONE together with the Aastra mobile client and our corporate communications suite takes Unified Communications a step further by adding true mobility. This combination leads to full freedom for employees to choose how and where to perform their duties irrespective of device and geographical location. Additionally, the management solution of MX-ONE is based on industry standards that IT departments are familiar with, making it easy to integrate into the corporate IT infrastructure.

With the release of MX-ONE 5.0 the customer can benefit from native UCC through the integration of the BluStar ecosystem end-points, such as the BluStar 8000i Media Desk Phone and the BluStar client portfolio. A new SIP user licensing model, with multi device/licensing possibility, is introduced to support the BluStar ecosystem of endpoints.

Aastra's BluStar ecosystem provides businesses with a comprehensive, end-to-end UCC solution with a versatile client that can be used across devices. BluStar offers a natural collaboration experience and fully embraces trends such as personal video and increased use of smart phones and tablets in the workplace. It helps users work together more effectively, e.g. on the fly video conferencing with remote workers, road-warriors and teams working on the same project to exchange ideas more easily and collaborate more efficiently. Aastra's focus across the portfolio is a unified user experience, facilitating adoption with a consistent look and feel, regardless of the device used.

The Management solution continues to be improved with its single point of entry approach, offering an efficient way of managing the system. Additionally, this release features full virtualization support for the MX-ONE TS communication server and UCC applications offering HW consolidation and enhanced resiliency options.

1.1 Scope

This document provides a high-level description of the Aastra MX-ONE solution. It includes a brief description of the system components, the network architecture, and external interfaces together with general feature descriptions for MX-ONE.

In this context, the term user is defined as an end-user within the enterprise that uses MX-ONE for daily communication. The user can access MX-ONE through several different terminals or applications. Thus, each user can have several extensions, for example, one extension for fixed telephony and one for mobile telephony.

Throughout this document there will be references to other documents that provide more detailed information about different subjects.

For information about how to implement MX-ONE in your IP network, see the description for *MX-ONE SYSTEM PLANNING*.

For information about telephony features and capacity data, see the description for *FEATURE LIST* and see the description for *CAPACITIES*.

This document contains the detailed information regarding MX-ONE Telephony System that was previously found in MX-ONE Telephony System Description, 13/1551-ANF 901 43 (which is phased-out).

1.2 The LIM Concept

The well known **Line Interface Module (LIM)** building block, which was a combination of call control element controlling its own dedicated hardware and telephony devices, has changed significantly with the introduction of the multi-gateway per server architecture with MX-ONE Telephony System V.4.0.

In the MX-ONE Telephony System, the LIM is now is divided into two main components. The call server component that takes care of the signaling is called **Telephony Server**, which is a Linux based call control software that resides on a standard Intel based PC architecture with processor unit, hard-disk, memory (RAM), input/output devices and connects to the LAN via NIC cards. The second component is called the **Media Gateway**, which contains an VoIP resource board and the physical interfaces towards TDM subscribers, public networks and Auxiliary devices. The media gateway provides circuit switch connections for legacy devices (ATS, DTS, and DECT) and the PSTN (ISDN, CAS, DPNSS and analog) and also houses the DSP resources for handling tones, conferencing, packet switching towards IP Phones (SIP and H.323) and to convert media between different protocols.

The MX-ONE Telephony System is developed on a modular concept using the IP infrastructure and server technology to further enhance the attractiveness of the well proven distributed architecture. A Telephony Server can now control up to fifteen media gateways for enhanced scalability, thus removing the one to one relationship that existed between call control and media gateway (a.k.a. LIM concept).

Although the term LIM is still referred to in this and other parts of the MX-ONE 5.0 documentation, it will be gradually removed from the documentation to avoid confusion with the multi-gateway concept.

Until the term LIM is removed, it shall be understood to mean the combination of the Telephony Server plus one or more media gateways. This terminology change does not in any way impact the system's ability to support existing installations with one to one server/media gateway configurations.

1.3 Glossary

For a complete list of abbreviations and a glossary, see the description for *ACRONYMS, ABBREVIATIONS AND GLOSSARY*.

2 System Overview

This chapter gives a high-level description of the MX-ONE solution.

2.1 General

MX-ONE consists of a set of integrated modules, forming a complete communication solution for an enterprise.

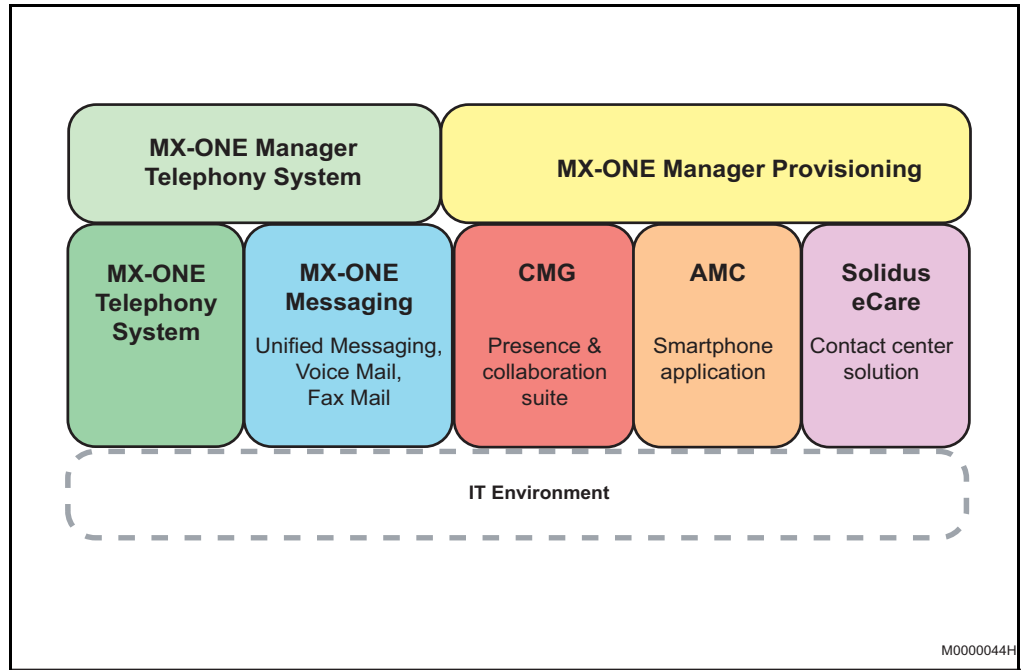


Figure 1: MX-ONE modules

MX-ONE comprises:

- MX-ONE Telephony System, the IP- and server-based foundation and major building block
- MX-ONE Manager Telephony System, a web-based application for configuring and managing telephony functions in MX-ONE
- MX-ONE Manager Provisioning, a web-based application for managing users and extensions in MX-ONE. MX-ONE Manager Provisioning provides a single point of entry for managing user and extension data in MX-ONE Telephony System, OneBox, CMG, and AMC Provisioning Server.
- Contact Management (CMG) is an advanced contact management system for telephony and provides applications for attendants and end-users.
- OneBox provides features for voice mail and fax mail, including Unified Messaging.
- Aastra Solidus eCare, a suite of applications and services for server-based contact centers.
- Aastra Mobile Client, a smart phone application that makes the phone an integral part of MX-ONE Telephony System with a rich set of MX-ONE application features.
- Aastra BluStar Client, a Unified Communications IP SoftPhone client application for PC, iPad and iPhone that enables corporate users to make and receive voice and video calls over SIP and to interact with other users and Solidus eCare agents through IM (Instant Messaging), e-mail and voice.

2.2 MX-ONE Telephony System

MX-ONE Telephony System provides business class telephony features. It performs call control, call-signaling, and media transcoding and conversion functions.

MX-ONE Telephony System can, in various configurations, support IP, mobile, analog, digital, cordless, WiFi, and CAS extensions.

The system can consist of up to 124 LIMs, where each LIM consists of a telephony server plus one to fifteen media gateways. For an explanation of LIM, see section 1.2 The LIM Concept.

MX-ONE Telephony System has the following capabilities:

- Acts as SIP server, H.323 gatekeeper and a media gateway
- Acts as a gateway towards private and public telephony networks, both fixed (PSTN) and mobile (PLMN)
- Offers a wide range of PBX telephony features.

2.3 MX-ONE Manager Telephony System

MX-ONE Manager Telephony System is a task-oriented web-based application providing functions for configuring and operating MX-ONE, for example:

- Performing setup of MX-ONE
- Managing media gateways
- Managing routes
- Managing attendants
- Managing groups, number plans, common categories, and service profiles
- IP phone and IP client provisioning

The screenshot displays the MX-ONE Manager web interface. The top navigation bar includes 'Initial Setup', 'Number Analysis', 'Telephony', 'Services', 'System', 'Tools', and 'Logs'. The user is logged in as 'aastra'. The main content area is titled 'Full Setup - Walkthrough - Step 2 / 29' and provides instructions for setting up media gateways. Below this, there is a table of configured Media Gateways (MGWs).

MGW	Type	Control IP	BPOS	Media IP	Subnet Mask	Default G/W
2A	MGU	10.105.69.235 (0)		10.105.69.245	255.255.255.0	10.105.69.1

Figure 2: Manager Telephony System

For more details, see the description for *MANAGER TELEPHONY SYSTEM*.

2.4 MX-ONE Manager Provisioning

MX-ONE Manager Provisioning is the user and extension management application in MX-ONE, providing a single point of entry for managing user and extension data in MX-ONE Telephony System, OneBox, CMG, and AMC Provisioning Server.

MX-ONE Manager Provisioning also provides functionality for example:

- Managing administrator accounts
- Adding subsystems, for example, MX-ONE Telephony Servers and CMG servers.
- Importing and exporting user and extension data
- Performing backup of user and extension data
- User account administration, for example, unlocking users.
- End-user self-service.

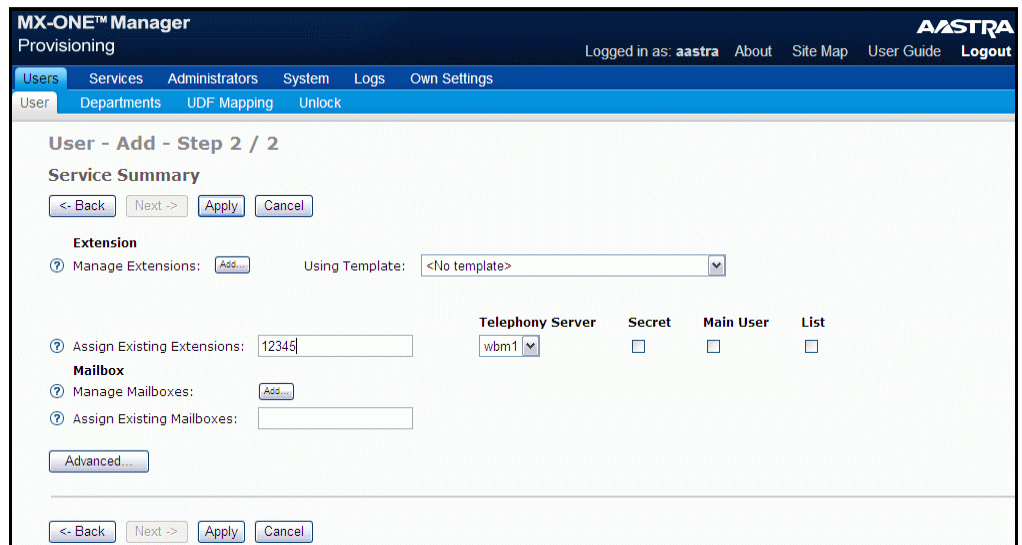


Figure 3: Manager Provisioning

For more details, see the description for *MANAGER PROVISIONING*.

2.5 MX-ONE Manager System Performance

Manager System Performance (MSP) provides simplified measurement and analysis of performance data from MX-ONE. It also gives the

MX-ONE administrator information about the overall performance of trunks, routes, operators, individual extensions and common system resources.

- Graphical interface for managing and viewing the MX-ONE performance.
- Easy to read and customize traffic information from the MX-ONE.
- Traffic analysis of major MX-ONE components, including Radio Base Stations and IP extensions.
- Multiple MX-ONE nodes can be managed.

2.6 MX-ONE Manager Availability

MX-ONE Manager Availability provides fault and performance management for MX-ONE servers and applications, that is, MX-ONE Telephony System, OneBox as well as the Linux and Microsoft Windows servers they run on.

Manager Availability is based on selected BMC ProactiveNet Performance Management components and prepared for integration with existing SNMP-capable management frameworks, therefore providing the benefit of leveraging customers' existing skills, processes, and management framework investments.

2.7 Aastra CMG Application Suite

CMG is a part of the UCC suite for MX-ONE. The CMG product portfolio provides a mix of communication tools for attendants and office users, including advanced call handling, activity and availability management as well as speech services.

The suite is developed from a user perspective and is a user-friendly contact management application that integrates well with MX-ONE, which enables the users to choose the most suitable way of communicating; via phone, video, conferencing, SMS or IM. It is a scalable system that can have virtually an unlimited amount of users and attendants. Using SIP standards, it can connect to several systems at the same time, either centrally or distributed over several sites.

Aastra CMG is a complete collaboration solution contributing to both Business users and Attendants with benefits for both target groups.

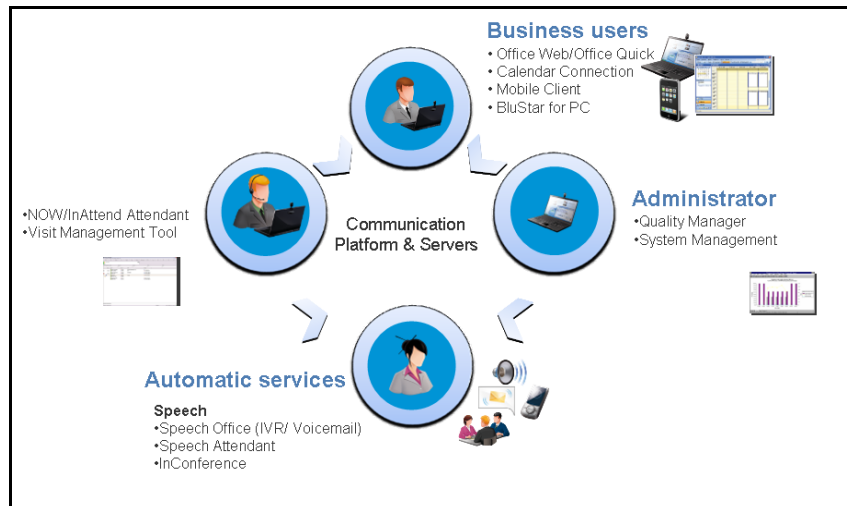


Figure 4: CMG Application Suite

CMG Application Suite consists of a number of end-user contact management applications for operating MX-ONE Telephony System.

The CMG suite is comprised of the following product areas:

- In Attend Advanced Attendant Console
- Presence and Availability Management
- BluStar for PC or OfficeWeb (a Web-based end-user portal)
- Calendar connection for integration with company calendar (Outlook, etc...)
- Visitor management.
- In-Conference meet-me conference solution

For more information on CMG, see the relevant CMG documentation.

2.8 Aastra OneBox

OneBox provides features and applications for voice mail and fax mail.

OneBox provides:

- A complete voice mail system that is fully integrated with the MX-ONE Telephony System via a SIP interface to be sure no calls are lost. Besides the MWI on the user telephones, extra notifications can be sent by e-mail or SMS to one or several devices per users.
- Full multi-level Auto-attendant functionality to allow inbound caller management as well as end-user voice mail box management.
- A web based user portal for managing voice messages and user preferences.

- An optional feature enabling full integration with major email systems (Outlook, Lotus notes and Groupwise) with advanced plug-ins for VM management.
- A voice intercept messaging (VIM) option which allows integration with the MX-ONE message diversion feature as well as the CMG activity setting function.
- A TTS (text-to-speech) option that allows user to listen to their e-mails directly over the phone.
- A fax server option that interacts with MX-ONE using standard legacy interfaces or using T.38 FoIP protocols.
- An ASR (Automated Speech Recognition) option that allows inbound callers to speak commands or for users to manage the VM box via speech commands.

Note: Some of the option mentioned above are optional and not licensed with the basic OneBox VM product. Activation of some of these optional features is accomplished by simply adding these features to the license file. Others, like OneBox fax-mail or ARS, may require installing additional software, which is residing on the media kit delivered with the system.

For more information please refer to the OneBox documentation.

2.9

Aastra Solidus eCare

Solidus eCare is an all-in-one, adaptive and flexible platform for UCC, mobility, contact center, Business Process Automation, analytics and reporting, as well as service and database integration. With the release of MX-ONE 5.0, including Solidus eCare 8.0, Aastra is continuing to build on supporting customers to transform their telephony-oriented call centers to true, two-way, multi-modal interaction hubs. This provides customers a choice of interaction methods by implementing multi-channel access capabilities, as well as more sophisticated APIs for business integrations.

New functionality introduced with Solidus eCare 8.0:

- BluStar agent application; new desktop application for agents
- A new web-services API for direct integration with configuration services
- A new router-based Web Callback option to perform Web Callback with SeC that has no dependence on the SeC Internet Suite
- Phone agent API improvements allowing applications development for phone agents such as: Logon/Logoff, Set Ready/Not Ready and Call Information when a service group call is allocated

Solidus eCare is empowered with Mobile Extension for Aastra communication systems to equip remote or roaming. Customers can be guaranteed access to the most appropriate agent - wherever they are located and on whichever communications medium (i.e., voice, chat, e-mail, SMS, or fax) they prefer to use.

Solidus eCare provides skills-based routing across these media, a single point of management and an integrated management information system across the contact center. The solution consists of software applications focused on the agent, management and customer self-service functions.

For more information on Solidus eCare, see the Solidus eCare documentation.

2.10 Aastra Mobile Client AMC

The Aastra Mobile Client is a smart phone application that is integrated with MX-ONE Telephony System to provide a Fixed Mobile Convergence solution. The installation and configuration is done Over-The-Air via a text message service and file download from the Aastra provisioning server.

The client's Graphical User Interface gives the user easy access to the different functions of the Aastra MX-ONE exchange, e.g. short number, conference, call waiting and message diversion. Similar to if the user was accessing/using the features from its desk phone.

As the mobile client can be reached via the regular PBX extension number, there is no need for the caller to know the public mobile phone number.

Through the AMC Mobile LCR, significant cost reduction can be achieved for the mobile user's long distance and roaming call setup.

Administration of the AMC users is handled over a dedicated web portal.

The AMC solution comes in two variants, AMC (single mode) and AMC+ (dual mode).

The basic functionality is the same but the AMC+ solution has an additional set of features to offer, e.g. Faster call setup, a-number/name presentation via data channel, directory search via LDAP and dual mode. The system architecture differs also between the two solutions.

Where the AMC single mode consists of a mobile phone application (client) that integrates with MX-ONE as a mobile extension. The AMC+ dual mode solution consists of both a mobile phone application (client) as well as a AMC Controller server (AMCC) that integrate with MX-ONE as a SIP extension.

For more detailed information about the AMC single mode solution see Astra Mobile Client 2.0 solution overview.

For more detailed information about the AMC+ dual mode solution see Solution Architecture Description AMC+ 2.0 with AMC Controller 2.0.

2.10.1 AMC, single mode solution

Feature set overview:

- Short number dialing & PBX features
- GUI for MX-ONE end-user features
- Mobile Least Cost Routing

Phone OS support:

- Android
- Symbian S60 3rd edition
- Symbian S60 5th edition (touch screen)
- RIM BlackBerry (4.x and 5.0 OS)

2.10.2 AMC+, dual mode solution

Feature set overview:

- Short number dialing & PBX features
- GUI for MX-ONE end-user features
- Mobile Least Cost Routing
- Dual mode (Wi-Fi/PLMN)
- Corporate Directory search (LDAP)
- Mobile Presence & Mobile IM (XMPP)
- Encrypted VoIP (STRP/TLS)

Phone OS support:

- iPhone iOS5
- Android 2.3 & 4.0
- Symbian S60 3rd edition
- Symbian S60 5th edition (touch screen)
- Symbian Anna & Belle
- RIM BlackBerry (5.0-7.0 OS)

2.11 Aastra BluStar for PC client

Aastra BluStar marks a new era in enterprise communications. It takes business communications to a new level across a choice of devices, providing consistent user experience by using video as the key driver. The term BluStar ecosystem refers to Aastra's portfolio of UCC clients, which consists of different software and hardware based components.

BluStar for PC delivers high-quality audio and HD video as well as access to a set of UCC features from a single client on the user's desktop. The direct integration with the MX-ONE Telephony System 5.0 provides the benefits of reduced cost and infrastructure complexity and secured high performance call functionality. With easy access to a set of UCC features from a single interface, the user experiences an intuitive tool that unifies voice communications with video, instant messaging, directory look-up, flexible search options and conferencing. By unifying all communication needs into one single client, the user can choose preferred communications channel depending on situation, even remotely outside the corporate network.

BluStar for PC brings the vision of a fully featured UC client a step closer. The BluStar for PC is an essential part of the BluStar ecosystem. It can be integrated with the CMG directory or any LDAP directory, e.g. AD to provide the benefits of presence, activities and directory handling and when interacting with the contact center.

The main features of BluStar for PC are the following:

- SIP softphone voice communications
- HD Video communications (peer to peer)
- Intuitive interface facilitates ease of use
- Powerful audio processing (echo cancelation, automatic gain control, codec support)
- Instant Messaging
- Contact integration (AD, LDAP, Personal Outlook Contacts, CMG)
 - Progressive directory search
 - Searches with local cache
 - Favorite list and grouping of favorite contacts
- BluStar click-to-call ("call to" links)
- Session Border Controller (SBC) enabling remote access
- Microsoft Lync integration
- Microsoft OCS and IBM Lotus Sametime plug-in integration
- Enhanced functionality with Jabra and Plantronics headset
- User license handled in MX-ONE 5.0

The BluStar for PC is a SIP based Softphone with a set of UC features that from a communication platform perspective is equivalent with any connected SIP terminal.

At start up, the client registers with the communication platform using SIP with or without password authentication. Media (RTP or MSRP) between the client and the remote end point is routed and negotiated using SIP via the communication platform during a call or Instant Message set up. Number translation is preferably done in the communication platform. When the negotiation is completed, direct media sessions are set up between the client and the re-remote end point. In some cases (remote end point located on PSTN or not a RTP capable terminal (i.e. analog phone)) media termination will be done through the communication platform or in a PSTN gateway.



3 MX-ONE Key Features

MX-ONE offers a comprehensive list of features and benefits for Enterprises, including:

System Architecture - The long proven scalability and flexibility architecture of MX-ONE supports the provision of multiple servers and multiple gateways system for customers to have a choice, rather than having a singular network for every location. Customers could have multiple servers networked together and supporting several hundred thousand users within their global voice network.

- **Scalability** - MX-ONE allows organizations to easily expand its capacity, add functionalities, modules, gateways, and easily upgrade the servers. From 100 to up to several hundred thousand users using a mixture of IP/SIP, mobile, digital, analog, and cordless extensions, organizations can benefit on the expand-as-you-grow principle of adding users in the system.
- **Flexible deployment** - MX-ONE is a highly flexible solution that supports a completely distributed model for deployment and design; Servers can be spread over and networked to create solutions for small and large customers across geographical locations or as a centralized system with the servers on a central location. With the flexible deployment and its scalable and modular approach to networking, MX-ONE thus also provides a globalized numbering strategy.
- **High Availability** - MX-ONE Telephony System offers high availability by supporting different types of redundancy to cater to different types and requirements of the Enterprises.

MX-ONE Telephony System offers far-reaching alternatives for High Availability:

Server redundancy delivering high availability:

- Home Location Register redundancy (HLR)
- 1+1 server redundancy and network redundancy
- N+1 server redundancy and network redundancy

Note: HLR redundancy can be combined with the other redundancy features. MX-ONE gateways (Lite and Classic) can also be configured in a redundant way, when MGU board is used.

There are two possibilities of **network redundancy** for an MX-ONE 5.0 Gateway:

- Bonding/Link failover
- Dual subnet

When using Server redundancy, network redundancy must be installed, and bonding is the preferred option.

A significant option from redundancy point of view in MX-ONE Telephony System is the server consolidation and Virtualization using VMware HA.

- **Virtualization** - With the advent of cloud computing technology, the cost of high availability, voice and data application hosting, content storage and service delivery will be reduced significantly.

Cloud Computing broadens the horizons across organizational boundaries of "reusability of IT capabilities" which is the very fundamental principle in cloud computing. To realize this computing

paradigm, MX-ONE has taken the bold steps in realization with its ability to employ Virtualization.

With MX-ONE it is now possible to run MX-ONE Telephony Server and its UC applications as virtual machines in a customer VMware environment. Thus, this capability has taken advantage of integrating real time communications as a service in the cloud environment. More importantly with MX-ONE using VMware Virtual Appliance, virtualization is supported and provides an alternative to MX-ONE's Resiliency and High Availability Solution.

- **True Mobility** - Mobility is a concept and a way of working, not just a method to forward your calls to your mobile device.

Aastra offers true mobility where a user is provided with the same services, the same access to corporate resources from any telephone, anywhere, connected to any network.

A user has one telephone number, one mailbox and is handled as an individual, irrespective of which terminal (or terminals) he is presently known to use by the system.

True mobility allows the user to log-on to any authorized telephone whether it is a local or remote IP device, an analog telephone connected to any network or a terminal or a DECT or WLAN infrastructure on the corporate premises or in public hot spots. True mobility is a native solution of the MX-ONE and no external servers or applications are required.

Aastra furthermore, offers the Aastra Mobile Client (AMC) which can be installed on terminals using Symbian S60 third and fifth editions, RIM Blackberry, Google Android and Apple iPhone devices, with no feature differences among the smartphones.

An in-house Fixed Mobile Convergence (FMC) solution allows seamless hand-off of calls between WiFi and cellular networks. Mobility is often costly, in particular for users that are travelling around the world. The AMC offers advanced routing functionality, GPRS call back and travelling SIM that can reduce the cost for international roaming with up to 90%.

- **SIP** - has become a leading signaling protocol for establishing real-time communications including voiceover-IP (VoIP) calls. SIP is Key to Accelerating Unified Communications Deployments in an organization. Aastra has taken advantage of using SIP in the development of MX-ONE. MX-ONE SIP extension and SIP trunk are available today and will continue to evolve to provide a simpler migration path to Unified Communication.

Protocol (SIP) trunks as a means of connecting Unified Communication Systems to the outside world are growing in popularity.

MX-ONE 5.0 SIP trunk will offer benefits over legacy PRI trunks in the future including:

- **SIP-Based endpoints** - MX-ONE supports a comprehensive portfolio of IP/SIP terminals. The portfolio consists of desk-phones, soft-clients, and cordless SIP-DECT. Aastra SIP terminals offer a high level of functionality and scope for design, utilizing embedded XML, which enables a customer to integrate the SIP terminal into applications and or the clients IT network.
- **SIP User licensing** - Maximizing value of the software and ensuring reliable access to the system in MX-ONE5.0 has taken the leap to introduce a new SIP user-licensing model, with multi-device support and multimedia licensing. The SIP user licenses are linked to users rather than to devices, making it far more flexible and cost efficient. "Multiple SIP end-points" means that a business user can have one directory number with a maximum of four active SIP end-points.
- **Comprehensive Suite of Unified Communication Applications** - UC is an increasingly important investment for organizations to keep on improving productivity and customer responsiveness while looking at reducing their IT costs. MX-ONE UC Solution provides a broad suite of UC applications that has an impact on business productivity and improves customer business performance by evolving and supporting enterprise business processes.

MX-ONE UC Solution has a truly user-centric approach and is based on open standards and open interfaces with mobility as an integral part allowing customers to make self-paced migrations. We have put together products in our MX-ONE UC Solution to achieve this, which includes BluStar 8000i Video desk phone, BluStar clients that work across multiple terminals (e.g. smart-phones), collaborative Web-based CMG user applications, the PC-based attendant InAttend, Unified voice messaging, the all-in-one contact center, and UC clients.

MX-ONE UC Solution enables activity and preference management that allow users to keep up-to-date quickly and efficiently on where they are, what device they are using and when and how they wish to be contacted. This allows employees in an organization to offer a personalized service to their customers.

- **Multimedia Support** - Video communication usage and the need for this technology are some of the fastest growing segments of the IT industry and which can offer the most benefit to customers nowadays. Video communication enables companies to utilize the use of it fast Internet to transmit video communication.

Video communication has evolved from an unreliable solution several years ago to a solid, pervasive technology that meets both the communication and cost containment needs of businesses.

The benefit of video communication is dramatic, not only providing benefits beyond business travel cost, but can also help enterprises modernize their operations and drive new revenue streams. Video communication and access to unified communications can also open up a host of opportunities for increased workforce productivity. New, flexible methods of communication and collaborations can be deployed with video services and applications.

MX-ONE has extended the advantage of Multimedia communication in its offering by fully integrating Aastra's BluStar ecosystem terminal, the **BluStar 8000i** and the different BluStar clients, the **BluStar for PC** when registered as SIP users with video privileges

- **TCO to Business flexibility** - For the organizations and businesses that are still using conventional telephony systems, the rising costs and lack of flexibility increasingly drives them to look for a new, more productive and cost-effective ways of communicating. The deployment of converged network communication delivering Voice over IP (VoIP) and UC applications is a natural and logical evolution. MX-ONE technology responds quickly to changing marketplaces and rapidly adopts technologies that improve business operations and processes for an organization to have a distinct competitive advantage.
 - **Simplified network infrastructure** - reduces costs by connecting IP/SIP extensions with IP/SIP phones or SIP clients over the WAN, seamlessly extending features to multiple sites through IP connectivity. IP trunking, instead of leased TDM circuits, enables a business to optimize network bandwidth and reduce network costs.
 - **Open server architecture** - MX-ONE runs on off-the-shelf operating systems and commercially available hardware servers.
 - **Business class telephony features** - MX-ONE with the most complete telephony feature offering for medium and large enterprises. No business has to make compromises on how to process critical customer calls.
- **Management Applications** - Dealing with complexity in today's dynamic communication infrastructures is one the many challenges faced by IT managers and support staff.

Increased reliance on communication solutions to drive profitability and achieve business goals has broadened the scope of IT and communication support issues while expanding the need for more reliable infrastructures and management tools. The recognition of

the issues has amplified the way MX-ONE management system works nowadays.

The single-point-of-entry management approach integrates well with today's IT management platforms and offers full control over your MX-ONE communication networks. MX-ONE offers the following web-based managers to fulfill the administrative, provisioning and performance monitoring on the same place for MX-ONE Telephony systems, UC applications and terminals.

- Manager Telephony system
- Manager Availability
- Manager Provisioning
- Manager System Performance

The Manager's functionalities are carefully developed and expanded to even further simplify and improve support for a greater portion of today's complex communication infrastructures. Enterprises can more effectively improve the value of their investments and better achieve their business goals and requirements.

4 What's new in MX-ONE 5.0

4.1 Virtualization

The MX-ONE 5.0 solution release extends to support larger customer multi-server environments, offers hardware consolidation and enhanced resiliency options and is aimed at customers who have a company-wide virtualization strategy for all IS/IT based applications.

The focused on consolidating the number of servers and limiting the number of physical servers necessary for both the MX-ONE Telephony System call manager and UCC application servers.

The integration focuses on VMware as the hypervisor technology and supports VMware value added features such as High Availability (HA) and Fault Tolerance (FT).

MX-ONE Telephony Server 5.0 can run in a virtualized environment, formally named Virtualization. Currently MX-ONE relies on VMware's vSphere 5.0 software suite as part of its virtualization solution

4.2 One Directory Number - Multiple SIP Terminals

From this version a user can have the same extension number on several SIP terminals of different types, with a maximum of 4 active terminals. From a licensing perspective, there is only one IP/SIP extension license per user, including support for one device. Several SIP devices can be connected to a user's extension number, If a user would like more SIP devices on his/her extension number, then "additional device" licenses for the extra terminals can be added. Furthermore, if a user would like to have video capabilities, then a video license per user can be added. This license is per user, rather than per video capable device, to allow for a user to have both BluStar soft clients and the BluStar 8000i and benefit from the same video license.

Supported end user terminals, any choice up to a total of four:

- Aastra 6700i desktop SIP terminal
- BluStar for PC
- BluStar 8000i Desktop Media Phone
- Remote extension over SIP, i.e. mobile (with or without AMCC)
- SIP DECT

Additionally, the enhanced multiple terminal service make it possible to have a combination of SIP end-points and generic devices, such as mobile extension and integrated DECT, with up to four devices registered to only one directory number. As an example, it would be possible to have one directory number with three SIP end-points and one mobile extension assigned. From the licensing perspective, the user would need only two extension licenses, one for SIP and one for ME, not four.

Note: For the teleworking scenario, the following limitation exists when a user works from a home office using an SBC for access to the corporate MX-ONE Telephony System: When multiple SIP devices per directory number is used, it is possible to have only one of those devices registered as a teleworking device (via the SBC) at any given time. An improvement is planned for SP1 Q4 where more than 1 but up to maximum 4 devices of different type can be registered through the SBC. It is of course possible to have one device registered over a VPN (e.g. BluStar for PC, for example) as well as a 6700i SIP phone via the SBC.

4.3 New SIP extension functionality with 6700i terminals

A number of new functions in SIP mode are introduced and supported with the 6700i terminal family:

- Dedicated key for diversion, with sub menus for follow me, message diversion, divert and individual "do not disturb"
- Shared Call line Appearance (SCA)
- Extra Directory Number (EDN) to offer similar functionality as Additional Directory Number (AND)
- Call Park Pool
- Intercom
- Group Do Not Disturb integration with MX-ONE Telephony System 5.0
- Possibility to set ring type for MNS (BLF) keys

4.4 Introduction of Unicode

The aim is to be able to display name as well as status information in the terminal displays, depending on type of terminal used. New firmware in the digital phones is mandatory to be able to use Unicode.

Unicode language support provides display of names and progress messages on terminals using other than Latin-1 code set which is required for countries such as Russia, Kazakhstan, China and Middle East. The system supports different languages for different users. The principle is that names will be forwarded and displayed transparently in the original language and progress messages in the language of the user.

The feature will be supported from the majority of terminals such as SIP, H323, DTS D4 (DBC223 and DBC225). SIP trunk will support Unicode. Unicode will not be supported from DTS pre D4 models, nor from DECT terminals or ISDN or H323 trunks.

Storing and processing names in the name data base is limited for name1 and name 2 lengths to 65535 unicode characters each. For correct printout format name 1 and name 2 are limited to 20 unicode characters each. The command for configure names impose also a limit of 20 unicode characters.

4.5 SIP trunk connection to ACS (Aastra Connectivity Server)

As of MX-ONE V.4.1 SP4, the SIP connection towards ACS was changed to a SIP trunk connection, rather than using a remote SIP extension, as was the case previously. This change also applies to MX-ONE 5.0. Furthermore, with MX-ONE 5.0 some net services are now available for this SIP trunk connection between MX-ONE and ACS to enable certain feature interactions between MX-ONE and the InAttend attendant suite. Configuration changes to the SIP trunk setting must be done in both MX-ONE Telephony System 5.0 and ACS to enable this enhanced feature interaction.

4.6 Supervisor with Silent Intrusion and agent parking

Parking a monitored call has been implemented as a new functionality between MX-ONE Telephony Server and Solidus eCare. It was not possible, when a Sec agent has been monitored, that the agent could put the call on hold for an inquiry call. This will be possible by now, the monitoring path will be lost for the time and the supervisor has to activate the monitoring again once the inquiry call is in speech state.

SeC agent is able to put the customer on hold to consult back office, while a supervisor is "monitoring" the agent by the intrusion feature in MX-ONE Telephony System.

4.7 MX-ONE Manager Provisioning 5.0

4.7.1 Operating Environment

The following lists the Operating Systems or versions of applications that should be used together with MX-ONE Manager Provisioning 5.0.

Manager Provisioning Server:

- SUSE Linux Enterprise Server 10 Service Pack 4

Manager Provisioning Clients:

- Microsoft Internet Explorer 6.0 or later
- Mozilla Firefox 3.6 or later

OneBox Unified Messaging:

- One Box 5.0 SP1
- One Box 5.0 SP3

- One Box 5.0 SP3 US1

CMG Server:

- CMG Server 7.5 SP2

4.7.2 **New functionality: Active Directory and Mapping of User-Defined Fields**

Since previously, Manager Provisioning can automatically import user records from Active Directory. Values of user attributes in Active Directory are copied to the Manager Provisioning user records.

It has also since previously been possible to define so-called user-defined fields in Manager Provisioning.

With this release, it is possible to configure Manager Provisioning to map its user-defined fields to attributes of Active Directory. The automatic user data import from Active Directory will include the mapped attributes accordingly.

4.8 **MX-ONE Manager Telephony System**

4.8.1 **Operating Environment**

The following lists the Operating Systems or versions of applications that should be used together with MX-ONE Manager Telephony System 5.0.

Telephony Server:

- SUSE Linux Enterprise Server 10 Service Pack 4

Manager Telephony Server Clients:

- Microsoft Internet Explorer 8.0 or later
- Mozilla Firefox 13 or later

SQL Server:

- PostgreSQL 8.1 (included with the Manager Provisioning media kit)

4.9 **MX-ONE Manager System Performance 1.0**

Manager System Performance (MSP) is a new application, part of the MX-ONE Management Suite. It provides simplified measurement and analysis of performance data from MX-ONE. MSP also gives the MX-ONE administrator information about the overall performance of trunks, routes, operators, individual extensions and common system resources.

4.9.1 Operating Environment

The following lists the Operating Systems or versions of applications that should be used together with Manager System Performance 1.0.

Manager System Performance Server:

- Windows 2008 Server (32bit), Windows 2003 Server R2

Manager System Performance Clients:

- Windows XP professional SP3, Windows Vista Business (32bit) SP2 or Windows 7 SP1(32bit)
- Office 2003 SP2 or Office 2007 SP2

SQL Server:

- MS SQL 2005, SQL 2005 express, SQL 2008, SQL 2008 express

4.10 MX-ONE Terminals (New Terminals Introduced with MX-ONE V.5.0 solution)

4.10.1 Aastra BluStar 8000i Desktop Media Phone

The Aastra BluStar™ 8000i Desktop Media Phone is a powerful desktop video conferencing and collaboration tool that is designed to enhance the way you communicate and collaborate.

Key features and benefits:

- Native HD video
- Outstanding HD audio
- Smart microphones
- In-person communication
- Business applications
- Intuitive communication
- Registered as a normal SIP extension with MX-ONE
- Support for LDAP corporate directory search
- Support for basic services, including hold, transfer, voice and video conferencing, fast forward, etc...
- Support for MNS keys in the "Favorites" list offering BLF functionality.

Note: When using the BluStar 8000i media phone with MX-ONE Telephony System, a video license is necessary.

When BluStar 8000i with the software 4.0.2 is used, there is no support to create the user unique configuration files automatically via Manager

Telephony System. These files have to be created manually according to installation instruction in CPI documentation.

When the software 4.1.0 is available it will be possible to create the user unique configuration files automatically.

4.10.2

SIP DECT 3.0

With this release we will also be introducing the new version of the Aastra SIP DECT family.

New functionalities introduced with this release are the following:

- Implementation of the new generation of the DECT standard, CAT-iq
 - Enabling handsets with CD level audio
- New range of base stations, supporting CAT-iq, GBit Ethernet and an interface for USB devices
- Combo base station - DECT CAT-iq and Wi-Fi (IEEE 802.11a,b,g,n)
 - No external antennas, ideal hotel solution
- Aastra XML interface enabling tighter integration with MX-ONE to offer an improved end-user experience
- New Aastra 650c CAT-iq DECT handset with wideband audio codec (G.722)
 - Superior audio quality

The following handsets are included with this release:

- 610d: Low end office version with basic features
- 620d: Business version with TFT color display and Bluetooth headset interface
- 630d: For industrial/care verticals, incl. alerting functions, protection class IP 65
- 650c: Business version, wideband audio, TFT colour, Bluetooth (commercially available Q3 2012)

A number of new functions are introduced and supported with the SIP-DECT:

- Diversion status: DND, FollowMe, external FollowMe, presence (absence reason (ie Lunch, Gone Home)
- Set Diversion via menu (the same menu provided to Aastra 6700i softkey phones)
- Profile [1..4] ; showing Personal number on an own line
- Only the telephone number and IPEI number needs to be configured in SIP-DECT OMM, enough to trigger a SIP REGISTER to MX-ONE

4.11 **Integration of Contact Centre Solidus eCare (SeC) 8.0**

The high quality multimedia contact center solution, SeC 8.0, is a part of Aastra UC suite with MX-ONE 5.0 Solution. SeC 8.0 includes the following functionality:

- BluStar Agent application
- Configuration web service API
- Router-based Web Callback
- Packet Browser installation for SeC including OAS, Internet Suite, Application Link and ELM i.e. all parts of SeC
- New backoffice/knowledge worker application, BluStar Expert

4.12 **Integration of Contact Management Suite (CMG) 7.5 SP2**

Aastra CMG is a part of the Aastra UC suite for MX-ONE 5.0 Solution. The CMG product portfolio provides a mix of communication tools for attendants and office users, including advanced call handling, activity and availability management as well as speech services.

The CMG 7.5 SP2 release is valid towards MX-ONE Telephony System 5.0 and besides fault corrections it includes the following new functionality:

- LDAP integration, in order to search the CMG directory using the BluStar for PC
- Calendar Connection SP4
- Possibility to list email addresses that can be excluded from synchronization

4.13 **InAttend 1.0 SP1**

The InAttend is a scalable attendant console based on open standards. It is the core application in Aastra's attendant offering and an essential part of the Aastra UC suite. In addition to advanced call handling, InAttend offers powerful search options, calendar integration, Microsoft OCS/Lync and IBM Lotus Sametime presence integration and all necessary information for efficient call handling.

The InAttend 1.0 SP1 release is valid towards MX-ONE Telephony System 5.0 and includes the following new functionality:

- ACS SIP trunk connection to MX-ONE Telephony System, allowing the following services to be executed:
 - Diversion bypass
- Chinese localization
- Caller identification, enhanced information of incoming calls entries in the Busy Lamp Field
- Configure favorites for the WebPanel
- Timezone support

For further information, refer to the product documentation and release notes available on the Astra knowledge base.

4.14 **Aastra BluStar for PC 2.0**

Aastra BluStar for PC 2.0 delivers high-quality audio and HD video as well as access to a set of UC features from a single client on the user's desktop.

The following features are delivered in BluStar for PC 2.0:

- Intuitive user interface for ease use of communications
- SIP softphone
- HD Video communications (peer to peer)
- Professional audio processing (echo cancelation, automatic gain control, codec support)
- Instant Messaging
- Contact integration (AD, LDAP, Personal Outlook Contacts, CMG)
 - Progressive directory search
 - Searches with local cache
 - Favorite list and grouping of favorite contacts
- BluStar "call to" link support
- Session Border Controller (SBC) support enabling remote access
- Microsoft Lync integration
- Microsoft OCS and IBM Lotus Sametime plug-in integration
- Integrated call control functionality with Jabra and Plantronics headset
- User license handling controlled in MX-ONE Telephony System

4.15 Unified Messaging OneBox 5.0 SP3 US1

OneBox SP3 US1 is a correction package for OneBox 5.0 SP3. There are no new features included with this correction package.

For further information, refer to the product documentation and release notes available on the Aastra knowledge base.

5 Telephony Features

This section describes some prominent features supported by MX-ONE Telephony System. For more information about the telephony features, see *FEATURE LIST* and *CAPACITIES*.

5.1 Extension Features

5.1.1 Extension Monitoring

Extension monitoring is an important feature for two main applications, Manager-Secretary and Team-working (key-system emulation).

The ability for a secretary to monitor the phones of one or more managers or for members of a team to monitor all members is enabled with the monitoring feature (MNS). It is possible to assign specific monitoring keys on IP phones and digital phones where the secretary and the manager can see the status (free, ringing, parked, or busy) of each others phones, indicated by a Light Emitting Diode (LED).

The manager or secretary can also use the programmed monitoring key to call the monitored party. Many configurations can be formed where individual secretaries can monitor the phones of one or more managers, as described in the figure below. In this example, the phone of Manager 1 is monitored by the phones of Secretaries 1 and 2, whereas Secretaries 2 and 3 are monitoring Manager 2. Secretary 2 then acts as backup to the other two secretaries.

The monitoring keys for Secretary 2 can have delayed ringing, giving time for the other secretaries to answer the call.

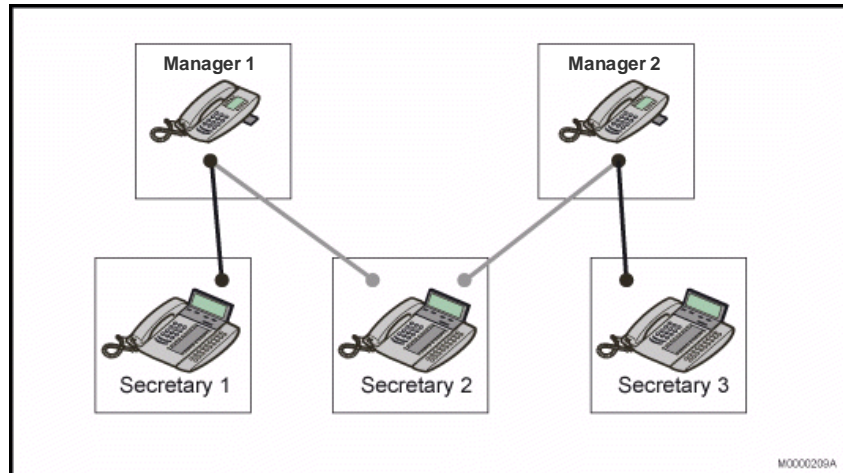


Figure 5: Example of a Manager-Secretary Solution

5.1.2 Parallel Ringing

The Parallel Ringing service provides the user with simultaneous ring signal on up to three predefined answering positions for an incoming call to the user. When the user answers the call, the call is directed to the extension where it has been answered.

However, the extensions might not ring at the same time depending on their type.

Parallel ringing supports analog, digital, CAS, cordless, IP, IP DECT, Wi-Fi and remote extensions (emergency extension and group number are not permitted). Any combination of the extension types may be used to define a parallel ringing list, but it is only possible to define one remote extension per list at any position.

For more information, see the operational directions for *PARALLEL RINGING*.

5.1.3 Personal Number

A Personal Number is an extension directory number with a terminal of any type (not ISDN S0) assigned, providing system users (voice extensions) with different possible answering positions for the incoming calls. Every personal number can have up to five lists where each list can contain up to ten different answering positions. The answering positions are selected depending on the settings for the user, for example, the user can be set to be at the office or to be at home. The settings of the personal number can be done manually by the user or automatically depending on calendar and location information stored for the user.

New in MX-ONE 5.0 is that a CTI group can be stored as a number in the personal number list.

For more information, see the description for PERSONAL NUMBER.

5.1.4 **Distribution Using Personal Number List**

Both the manager and the secretary can configure the diversion of incoming calls to the manager. While the feature is active, the calls to the manager is deflected to the secretary's phone, and a LED associated to a key on both the manager's and secretary's phones indicates that the service is active.

The service can be activated and deactivated by both managers and secretaries by pressing the previously defined key. When the service is active, the LED associated to a key is turned on at both the manager and the secretary phones (digital and IP phones are supported but cannot be mixed, meaning that the manager and the secretary must both have digital phones or both have IP phones). When the service is inactive, the LED associated to a key is turned off at both the manager and the secretary phones (digital and IP phones are supported but cannot be mixed). The service can be directly deactivated by pressing the previously active key or indirectly deactivated by activating another personal number list.

5.2 **Attendant Features**

In MX-ONE, the attendant can handle all types of calls and assist users to activate and deactivate features. The attendant terminal is normally the InAttend application but older types, e.g. NOW from the Contact Management (CMG) application suite and hard consoles (OPIs) can still be used. Calls are distributed among the active attendants. If there are no free attendants available, the calls are placed in queue. A queued call is distributed as soon as an attendant is available.

The attendant functionality is provided by a separate server, the Aastra Connectivity Server (ACS). The ACS is connected with SIP to the MX-ONE Telephony System. The ACS provides queuing and call distribution. The use of ACS and SIP connection makes it possible to offer InAttend also to customers with PBXs from different vendors but still wants a uniform attendant functionality.

For more information about NOW and ACS, see the respective user manual for *NOW* and *ACS*.

5.2.1 **Call Origin Groups**

To distribute incoming calls to an attendant that has the best qualifications to handle a specific call, the calls can be routed to different groups called call origin groups. An individual attendant can be affiliated to one or more call origin groups.

A call origin group is a combination of call origin type, route number, attendant call number, and customer number. Each call origin group can be of one or more call origin types.

The following example describes two call origin groups. Call origin group 1 receives external calls and calls that have been rerouted. Call origin group 2 receives internal calls and emergency calls.

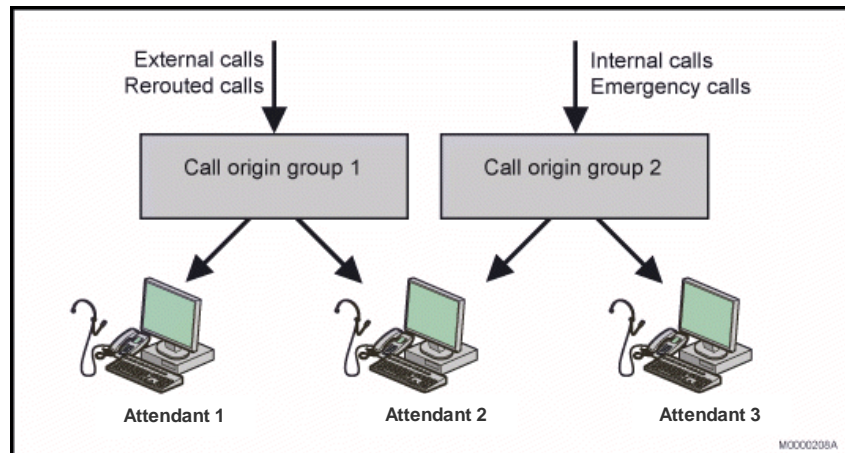


Figure 6: Call Origin Groups

Call origin group 1 distributes calls to attendants 1 and 2. Call origin group 2 distributes calls to attendants 2 and 3. This means that attendant 2 can handle all types of calls, whereas attendants 1 and 3 can only handle a limited number of call types.

If the customer group feature is used, it is possible to affiliate the attendant to a specific customer, see 5.5 Customer Group.

Another related feature is centralized attendant, see 5.4.1 Centralized Attendant.

5.2.2

Call Handling

An incoming call can be answered automatically, that is, the attendant does not have to press any key, and it is possible to play a welcome message, so the attendant does not have to repeat the same phrase.

The attendant has full control over both the incoming call and the extension to which the call should be extended. The PC screen of the InAttend application is split in a left and a right side, making it possible to see both connected parties.

If the called party is free, the attendant can choose to extend the call before answer or wait for the answer to announce the calling party before extending.

If the called party is busy with an ongoing call, the attendant can camp on the busy extension.

5.3 Group Features

Users can be grouped into different types of work groups. The users connected to a specific work group can use a number of group related features, such as the ability to pick up calls to other extensions in the group or use a common number.

For more information, see the description for *EXTENSION GROUPS*.

5.3.1 Group Call Pick-Up

The feature group call pick-up makes it possible for a member of a predefined group to pick up a call to any other member in the group when an incoming call is made to that member's extension.

Calls to one extension in a group can be answered by any group member. Each group can have four alternative answering groups and, if there are no calls to members in the group, calls to the alternative groups are answered with the same procedure.

5.3.2 Group Hunting

It is possible to form groups of users that work with similar tasks like help desks, information centers, and so on. Such groups are called with a common number. Incoming calls are routed to a free extension in the group, either with sequential hunting or evenly distributed.

If there are no free extensions, the call is queued to the group.

All extensions in a group keep their own private number and class of service. An extension can be a member of several hunting groups. An extension can temporarily withdraw from the group by activating Follow-me to its own phone.

Calls to a group from which all members have excluded themselves are diverted to the group's divertee position.

5.3.3 Group Do Not Disturb

It is possible to form groups of users that during certain periods must not be disturbed. The group do not disturb feature allows a specially categorized extension (master extension or extension with Group Do Not Disturb programming category set) to mark an extension group as group do not disturb, that is, calls made to extensions in the group are not signaled on the phone.

If any extension has any diversion activated or an individual divertee position, the call is diverted. The diversion feature has a higher priority than the group do not disturb feature. If the extension has no diversion

and the incoming call is a Direct In-Dialing call that has a category permitting rerouting, the call is rerouted to an attendant. If the extension has no diversion, and the incoming call is not a Direct In-Dialing call then, it shall be forwarded to an answering position defined for the group.

5.3.4 Call Center (CTI) Group

The Call Center Group (CTI group) is a class of service for the Automatic Call Distribution (ACD).

Extensions can be members in a call center group. In this case, the incoming calls to the group, both external and internal, are routed to an external contact center application using the Computer Supported Telecommunications Application (CSTA) interface and protocol. The contact center application selects a free extension, or if there are no free extensions, places the call in a queue. When there is a free extension, the call is routed to the selected extension.

5.4 Private Network Features

MX-ONE Telephony System can be included in a private network with other MX-ONE systems or MD110 systems or even systems from other vendors.

To use network services between all parties within a private network, the network must be homogeneous, which means that the connections between the exchanges use the same signaling system. Because the network services are not supported in a gateway scenario with transition from one signaling system to another. Connections using ISDN and H.323 signaling systems are considered a homogeneous network.

Systems 1 to 4 in the below figure are connected to each other using IP networking (H.323 with additional proprietary signaling). Systems 5 and 6 are connected to the network through leased lines using ISDN with QSIG signaling. All systems can have their own incoming and outgoing external lines to the PSTN.

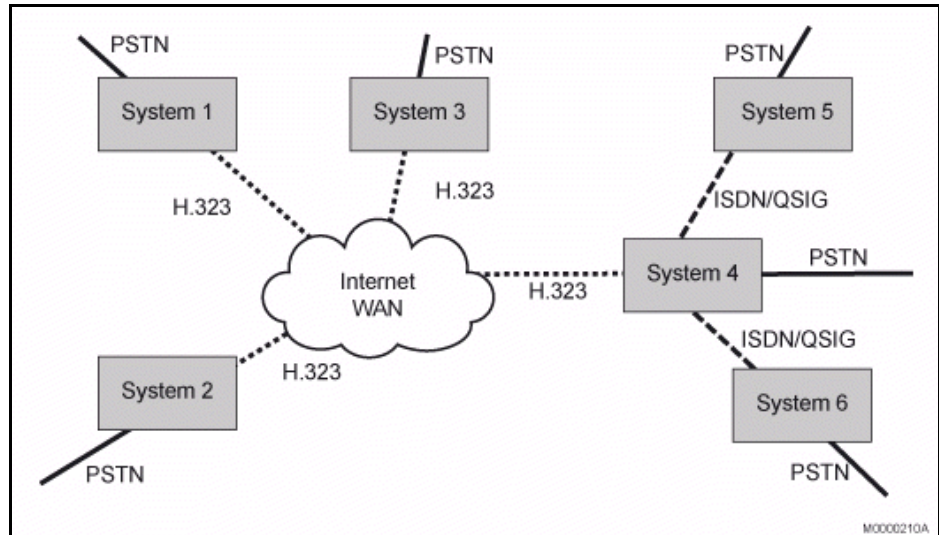


Figure 7: Private Network Features

The following documents include more information:

- Description for *NETWORKING*
- Description for *IP NETWORKING*

5.4.1

Centralized Attendant

Both local and centralized attendants can be used in a networked scenario. A centralized attendant is an attendant shared by several PBX systems. The advantage is that when the total traffic in a network is low, all traffic can be routed to a central point, thus minimizing the number of attendants. The feature can also be used in a global enterprise where all calls are routed depending of the local time of day to those systems where attendants are working.

In the example below there are three systems. System 1 has a full time attendant, system 2 has a local part time attendant whereas system 3 does not have any local attendant. Incoming manual calls in system 3 are always routed to the centralized attendant in system 1. Incoming manual calls in system 2 are when the local part time attendant is night switched, routed to the centralized attendant in system 1 whereas, when the attendant is day switched, the calls go to this local attendant.

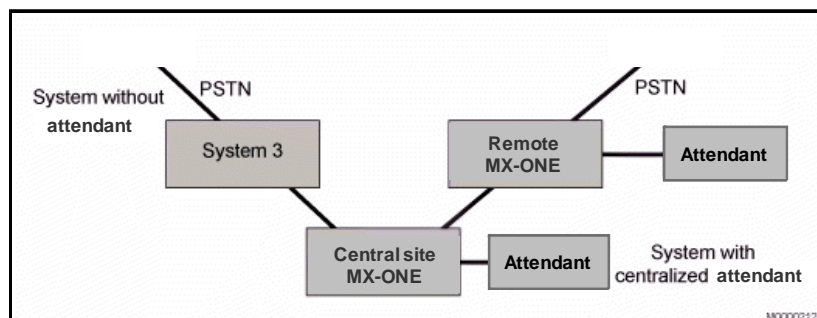


Figure 8: Centralized attendant

5.4.2 Least Cost Routing

The Least Cost Routing (LCR) feature enables MX-ONE Telephony System to select the most economical route for an outgoing public call. The system checks the dialed number to see if the private network can be used, and then routes the call to a PBX within the private network as close to the destination as possible.

The following three situations can occur:

- The dialed destination is an extension within the own system.
- The dialed destination is in the private network and can be reached completely through the private network.
- The dialed destination can be reached partly through the private network.

The user first has to dial an LCR access code to activate the LCR. The system is then ready to analyze the dialed number.

5.4.3 Alternative Routing

The alternative routing feature makes it possible to reach an external destination through different routes. A primary choice route can have several alternative routes to reach the external destination.

If the primary route is not available, the system tries to use the first alternative route and so on. If necessary, the system modifies the dialed number (add pre-digits or delete dialed digits or both) to suite the number to the numbering plan used in the exchange at the other end of the alternative route.

For an example where a user in system 1 wants to call a user in system 3, the system first tries to find the direct route (alternative 1), but if all connections in that route are unavailable, the system tries to connect to system 3 through system 2 (alternative 2).

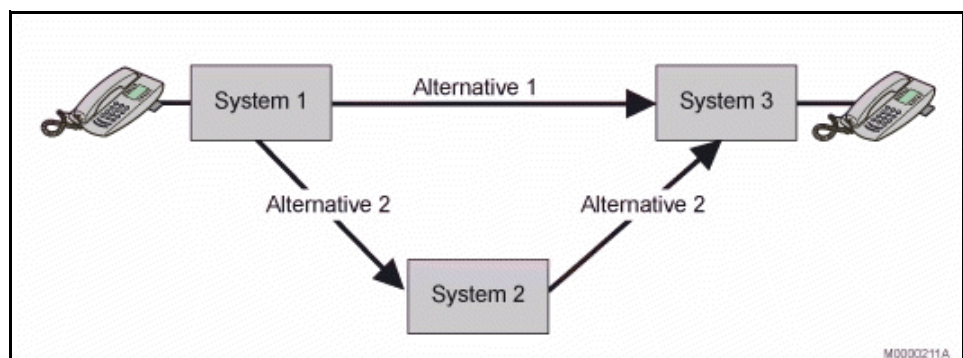


Figure 9: Alternative Routing

5.4.4 Network Features

In MX-ONE Telephony System, a number of common network features are supported. Typical network features are diversion, callback, intrusion, and call waiting (call offer). To use the network features fully, it is necessary to have a signaling system that can convey the necessary information.

Two groups of common network features are distinguishable, that is, system and user features (covering both extension and attendant features). The system features are used by exchanges to route calls in a controlled, rapid manner to the called destinations. A user can only affect user features, not system features.

For connections with ISDN, system features as well as user features are applicable. For connections with packet-based signaling, H.323, system features as well as user features are applicable.

The network features are divided in those using standard QSIG signaling and others using proprietary signaling. When MX-ONE Telephony System is connected to an MD 110, all features are available for both ISDN and H.323 signaling. Other types of signaling is also supported, for example, DPNSS.

5.5 Customer Group

An MX-ONE Telephony System can serve customers belonging to different enterprises that can be completely separated regarding telephony. The customers are arranged in one customer group, which can consist of a maximum of 250 customers. MX-ONE Telephony System only supports one customer group. This feature is also known as a multi-tenant group.

Each customer has its own virtual system. Customers within the customer group can have their own resources, such as routes and attendant groups, but can also have features for using common system resources within the customer group.

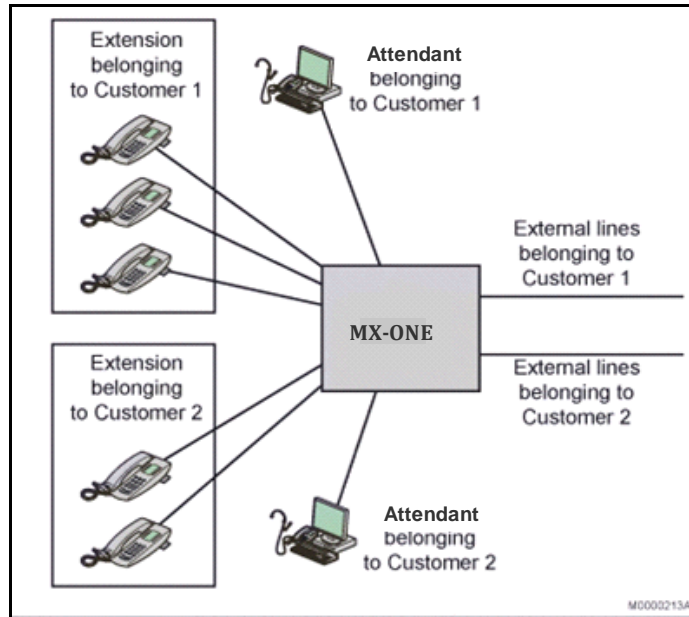


Figure 10:Customer Groups

A customer can be assigned to a group of attendants, a day or night service answering position, outgoing routes, and incoming routes.

The customer group feature makes it possible for the users in a customer group to, for example, reach the affiliated attendant group, by dialing a common attendant number. This feature enables subdivisions within an enterprise or enterprises to use MX-ONE independent of each other.

It is recommended that the different customers have separate incoming and outgoing routes, but this is not mandatory.

The customer group feature includes the following five major functions:

- Customer-dependent selection of attendant group for calls to the customer group's common attendant number
- Customer-dependent selection of centralized attendant, that is, customer centralized attendant, for rerouted calls and calls to attendant group
- Customer-dependent day and night service position
- Customer-dependent route selection
- Common diversion positions

5.6 Hospitality

The special Hospitality functions are provided for the Hospitality industry. The functions are provided within the following areas: guest check-in, guest rooms and Service quarters.

For guest check-in the following functions are provided:

- Information about the guest can be entered. The information can be language, VIP-status, credit card information, room number, and so on.
- The guest can choose if name presentation will be allowed or restricted for inter-room calls.

For guest rooms, the following functions are provided:

- The rooms can be either vacant or occupied.
- The rooms can be equipped with various types of terminals.

For Service quarters the following functions are provided:

- The guest's name presentation restriction will never be taken into consideration when the Service quarter is in contact with a guest, that is, the name of the guest will always be displayed at the Service Quarter Terminal.
- The information that is entered when a guest checks in will be displayed at the Service Quarter Terminal.

For more information, see the description for *HOSPITALITY APPLICATION*.

5.7 VoIP Recording

With active VoIP recording, the recording system will order the terminal to make a copies of received and sent RTP or SRTP packets and forward those to the recorder, preferably as two logical streams where received and sent packets are separated. Certain features of advanced VoIP recording systems are dependant on the receipt of two separate voice streams.

Recording policies are defined in the recording systems, i.e. record all calls (Total recording), selective recording based on e.g. call origin and record on demand based on user input.

Active VoIP recording relies on integration with the PBX as the recording system must know the actual IP address of the users terminal when ordering a terminal to start recording. Furthermore, if encryption is used, the recording system must get the encryption keys. The integration is normally using SIP between the recorder and the terminal and CTI between PBX and recorder.

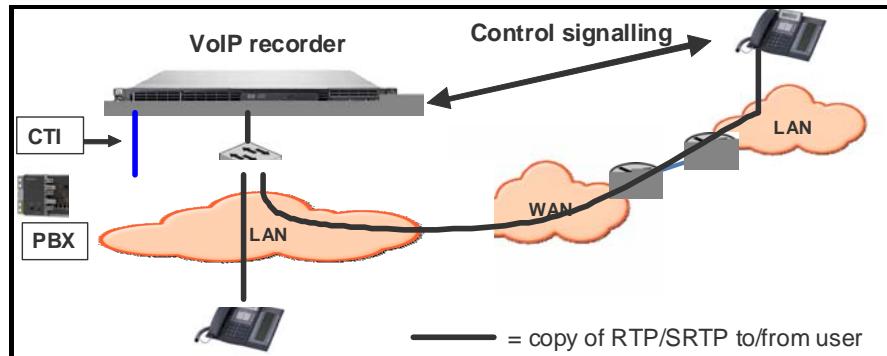


Figure 11: Active IP recording

5.8 Recorded Voice Announcement

The Recorded Voice Announcement (RVA) feature allows recorded voice announcements to be provided to a calling or connected party to inform of the status of the call in various traffic cases, for example when:

- the call is diverted
- the call is in queue
- the call is parked
- the call is answered by a PBX-operator

The types of calling or connected parties that receive RVAs are, generally speaking, external lines and extensions, with some exceptions. For example, for diversion, group hunting, Automatic Call Distribution (ACD), call to an operator or to an extension, voice announcement can be provided when the call has originated from an extension, from a public trunk line, or from a tie line. Vocal guidance is a service that can be provided for analog (ATS), digital (DTS), cordless (CXN), and certain mobile (RXN) extensions, a recorded voice announcement can be played when the user encounters certain traffic cases e.g. External Follow Me (ECF). Music on Hold can be provided for certain call cases towards group hunting and ACD groups.

5.9 Name and progress presentation

On terminals (telephones) equipped with display name information and progress (status) information is displayed. The MX-ONE Telephony System uses Unicode (ISO 10646-UCS-4) for both name and progress information.

Currently progress information can be presented in most Western European Languages as well as in Russian. The language to use can be configured for the system, but also for individual extensions

6

Architecture

MX-ONE comprises MX-ONE Telephony System that serves the core voice traffic, InAttend for the attendant services, BluStar for PC and CMG Application Suite for personal communication and preference setting, MX-ONE Manager for user, extension, and system management, Solidus eCare for business process integration and automation and OneBox for voice mail and fax mail.

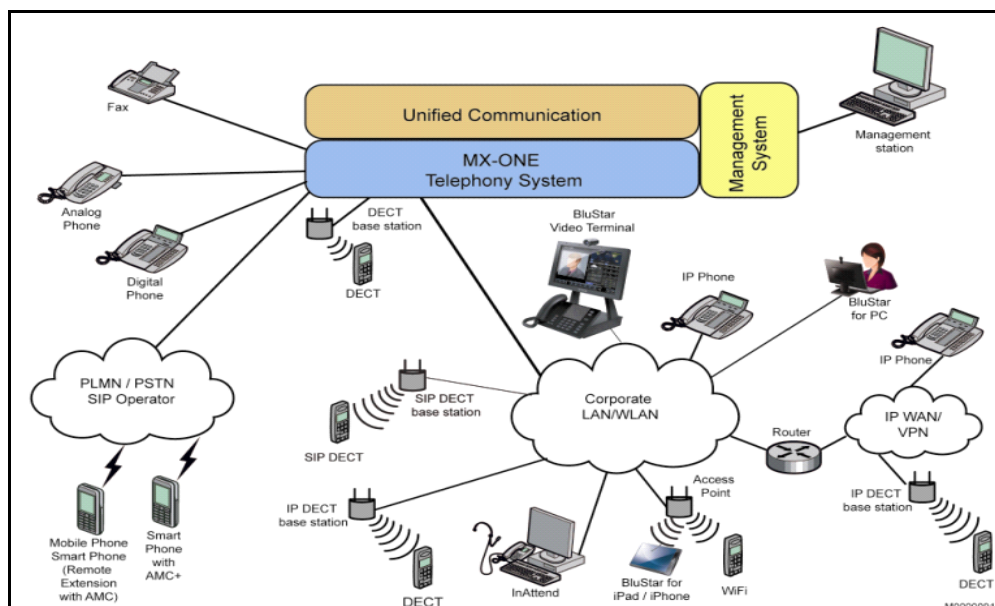


Figure 12: Architectural Overview

The following sections give an overview of the different components and their primary features in MX-ONE. For detailed information, see the description for each individual component.

6.1

MX-ONE Telephony System

MX-ONE Telephony System has evolved into a true Multimedia Communication System that can route and provide services, as applicable, also to media sessions like Video and Instant Messages.

MX-ONE, as all other communication servers, needs a high quality IP network, such as Ethernet, in order to operate and provide high quality communications. This IP network shall be designed according to the best practice in the market to accommodate high quality Real Time services over IP. MX-ONE Telephony System provides business class telephony features. It performs call control, call-signaling, media transcoding and conversion functions. MX-ONE Telephony System operates as an application in the customer's IT environment and is fully

supported for operation as a Virtual appliance in an Enterprise cloud. Services like voice and video running over packet networks needs a data link layer where the Quality of Service (QoS) capability is configured for low delay and low packet loss, parameters of great importance for high quality media sessions. The entities of LAN switching and IP routing are not part of MX-ONE Telephony System offering as such, but are rather a prerequisite for a functional solution.

MX-ONE Telephony System can support any combination of users with terminals like IP, mobile (cellular), analog, digital, cordless DECT, IP DECT, SIP DECT and Wi-Fi, as well as legacy interfaces like CAS and paging equipment. For connectivity to other applications and systems, the MX-ONE Telephony System can support any combination of trunking like SIP, H.323, ISDN including Q-SIG, as well as legacy interfaces like Loop Disconnect, Ground Start, E&M, CAS, MFC, DPNSS and DASS.

6.1.1 MX-ONE Hardware Architecture

MX-ONE Telephony System consists of logical modules, earlier called LIMs. The minimum setup is one server plus one media gateway, that is, the same as the old LIM. A telephony server can control from one to fifteen media gateways.

6.1.1.1 Media Gateways supported in MX-ONE 5.0

The following media gateways are compatible with MX-ONE 5.0

- MX-ONE Lite (3U with MGU) *)
- MX-ONE Classic (7U, 19" with MGU) *)
- Media Gateway Classic (7U, 19" with LSU-E) **)
- Media Gateway Classic LBP23 (7U, 19" with LSU-E) ***)
- Media Gateway Classic Stackable (with LSU-E) ****)
- Enterprise Media Gateway, EMG (1U) *****)
- Compact SM (2U combination of EMG and MX-ONE Server) *****)

*) Only MX-ONE Classic and MX-ONE Lite are available for new sales in MX-ONE 5.0.

***) Media Gateway Classic with LSU-E only exists in systems upgraded from earlier versions of MX-ONE.

****) LBP23 only exists in systems migrated from TSW with duplicated control system. These systems have limited capacity.

*****) Media Gateway Stackable is service stopped and is handled according to 'Older building practice' paragraph below.

- *****) EMG and Compact SM will be service stopped 2013-06-30 and are thereafter handled according to 'Older building practice' paragraph below.

6.1.1.2

Older building practice

Older, service stopped (Stackable and pre-stackable), building practices have limited support in MX-ONE 5.0 according to the following principles:

- Service stopped HW is permitted on an "as is" basis. It is, however, strongly recommend-ed to move extensions on this old HW to more modern HW to avoid any risks.
- Should the service stopped HW fail or break-down, they can no longer be serviced and therefore the customer must transfer the affected subscribers and trunks on this chassis to more modern HW.
- Should existing older boards (stackable and earlier) be maintained in the installation, they may not be placed into the new building practice, but left in existing cabinets or moved to a stackable chassis. However, the preferred solution is to replace them with today's modern HW that offer equivalent functionality with less foot print and power dissipation.
- MGU and ASU-E can not be placed in older cabinets.
- It is no longer possible to repair or order spares parts for phased-out HW, therefore if an old line board break down, the users must be moved to supported HW.
- New Media Gateway HW must be purchased for replacement of old interface cards, as repair or spare parts no longer exist for phased-out HW.

6.1.1.3

The following types of servers and gateways are supported in MX-ONE 5.0

- Standard Server (HP/Dell)
- Software Only (own operating system including Linux support subscription and own choice of server fulfilling the minimum requirements)
- Aastra Server Unit Embedded (ASU-E) (integrated server board)
- Aastra Server Unit (1U ASU)
- MX-ONE Server Unit (1U ESU)
- ESU (integrated server board, for migration)

Server/Gateway combinations

- One to fifteen media gateways can be connected to one standard server.
- ASU/ASU-E supports up to fifteen media gateways with a total of max. 4000 users.
- ESU/MX-ONE Server only supports one media gateway, except for upgraded or migrated systems where it can support two media gateways.
- EMG and Compact SM must have one designated server each, and cannot exist in a multiple gateway scenario.
- MGU and LSU-E based media gateways cannot co-exist on the same server.

Table 1 Server Options

Gateway	Type of system	HP	ESU	MX-ONE server	ASU-E	ASU
EMG	Traditional	Yes	N/A	Yes	N/A	Yes
Compact SM	Traditional	N/A	N/A	N/A	N/A	N/A
MX-ONE Lite	Traditional	Yes	Yes	Yes	Yes	Yes
Media Gateway Classic Stackable (LSU-E)	Traditional	Yes	Yes ^{a)}	Yes	No	Yes
Media Gateway Classic (LSU-E)	Traditional	Yes	Yes	Yes	Yes	Yes
MX-ONE Classic (MGU)	Traditional	Yes	Yes	Yes	Yes	Yes
MX-ONE Lite	Multi	Yes	No	No	Yes	Yes
Media Gateway Classic Stackable (LSU-E)	Multi	Yes	No	No	No	Yes
Media Gateway Classic (LSU-E)	Multi	Yes	No	No	Yes	Yes
MX-ONE Classic (MGU)	Multi	Yes	No	No	Yes	Yes
MX-ONE Classic (MGU) ^{b)}	Multi (max 2 GW)	Yes	Yes	Yes	Yes	Yes

a) Integrated fan board is required.

b) A special case for migrating customers.

6.1.1.4

Deployment of MX-ONE Media Gateway

In MX-ONE, each media gateway can function as an autonomous IP-PBX if it is complemented with a dedicated server, for example, MX-ONE Lite equipped with an ASU-E.

A number of media gateways can also act as common VoIP and TDM interfaces and resources without a dedicated server. In this case, the servers forming the system are configured with the number of media gateways that are required to build larger systems from a capacity perspective.

With this flexible deployment approach, the capacity of a MX-ONE telephony system can be scaled from approximately 100 to several hundred thousand extensions in normal traffic conditions. In a large scale MX-ONE system deployment, the Servers and media gateways are interconnected through a LAN. An MX-ONE Telephony System can consist of up to 124 (LIM) servers in a multi-server system. MX-ONE offers high availability by supporting both server redundancy and network redundancy. For more information, see chapter 5.5 High Availability.

Below, several terminologies are explained to clarify MX-ONE media gateway deployment scenario to build a highly scalable MX-ONE Telephony System.

Single gateway MX-ONE system

A single gateway is defined as a media gateway, either MGU based or LSU-E based, with its own Telephony Server, i.e. a single media gateway MX-ONE system consists of a Telephony Server and a media gateway. Refer to section 6.1.1.1 Media Gateways supported in MX-ONE 5.0 and Table 1 Server Options for various media gateway types and valid server options.

Multi-gateway MX-ONE system

Multi-gateway is defined as a number of single gateways, either MGU based or LSU-E based, with a single Telephony Server for call control and management.

An MX-ONE system can comprise of several multi-gateways with different media gateway types. However, it should be mentioned that a multi-gateway must contain same types of media gateway (all MGU based or all LSU-E based).

Remote Gateway

A single gateway or media gateway(s) in a multi-gateway system can be placed in remote location(s) (over the WAN).

In case, a single gateway system, i.e. with its own Telephony Server, is deployed remotely, it is rather referred to a remote system node which is a part of a large MX-ONE system configuration. This remote system node is a complete and independent MX-ONE system supporting all features and services without any restrictions due to the geographical distance.

A remote gateway is usually referred to a media gateway which belongs to a multi-gateway MX-ONE system (e.g. located in the main HQ site)

but placed in a remote site without local server. It is a cost effective way to facilitate a remote MX-ONE node, for instance to build a branch node. However, there are several issues that must be taken into consideration for the remote gateway deployment.

- Bandwidth
- If WAN access link for the remote gateway is provided with narrow bandwidth connection, the bandwidth may not be enough to carry incoming and outgoing media traffic depending on the system size (number of users).
- Care must be taken that there is enough bandwidth for a worst case scenario.
- Latency
- The latency or packet delay in terms of the call and media control orders and responses is crucial when the media gateway is located remotely. The max. round trip delay should not exceed the allowed time limit for time-out (normally < 100 ms is required) for proper operation.
- Availability/redundancy
- When media gateway loses contact with the Telephony Server (e.g. due to the WAN failure), the media gateway service will be disrupted as no PSTN fail-over or local survivability is supported in MX-ONE 5.0.
- It is not straight-forward to support redundant network access with media gateway located remotely and even if a standby link can be assigned, the down-time can be substantial in the presence of WAN failure.
- Emergency (SOS) call service
- To enable the emergency call service, procedures described in relevant operation directions must be followed carefully. It may not be possible to provide fully transparent services depending on the terminal types to dial the emergency number and the route access code defined to locate the right emergency center. For more details, refer to Operation Directions for *Emergency Calls, SOS Calls*.

Considering the issues listed above, a remote MX-ONE system (i.e. media gateway with own local Telephony Server) rather than remote gateway is strongly recommended. In particular, the local survivability for non-disrupted service is a crucial factor that motivates an MX-ONE system instead of remote gateway alone for branch node application.

6.1.2

Multi-Server Features

An MX-ONE Telephony System composed by several servers (previously called multi-LIM system) is in all respects a single PBX system. IP

extensions can register with any server of the multi-server system. With the new HLR redundancy feature, an IP extension does not need to be able to contact its home-server for registration although certain limitations still applies. At installation, each server is assigned a server number. Server 1 always runs Manager Telephony System (see 6.2.1 MX-ONE Manager Telephony System).

In a multi-server system one arbitrary server is assigned as master and is constantly monitoring the other telephony servers by a mechanism similar to that used in a Token Ring network. If one server fails or is unavailable, the system will notice this and continues to work correctly. If the master server fails, another server will take over this role. In a multi-server system, the IP infrastructure (LAN) is used for inter-server communication, both call control signaling and media. All signaling is sent between the Telephony Servers over the LAN using a proprietary protocol over Stream Control Transmission Protocol (SCTP). The media streams are sent over the LAN using the Real-time Transmission Protocol (RTP) protocol.

A group switch can also be used to send the media streams, but only between single media gateway servers. If both parties in the call are IP phones or IP clients, the media streams are sent directly over the LAN between the parties. Note that both end-points must use the same protocol. If one or both parties in the call are Time Division Multiplexing (TDM) based or an analog phone, an IP gateway is used to bring the party to the LAN. The media gateway has a built-in IP gateway connecting the Primary Rate Interface (PRI) trunk lines to the LAN. The Media Gateway Classic uses IP Line Unit (IPLU) boards as IP gateways to the LAN. An MX-ONE can be installed with a group switch, called GS. GS is placed in the same cabinet as a media gateway. The group switch only handles media and establishes the media connection between media gateways. The group switch is built up by Group Switch Modules (GSM). Each GSM can have 31 PCM lines connected to it. A group switch can have up to 8 GSMs, and it can be duplicated. The group switch cannot be used in multiple media gateway systems.

For more information, see *MIGRATING TSW/MD110 TO MX-ONE 5.0*.

6.1.3 Media Gateway Hardware

MX-ONE Telephony System supports several types of media gateways. This chapter describes the media gateways available for new deliveries.

6.1.3.1 MX-ONE Lite

The MX-ONE Lite media gateway is based on the MGU circuit board mounted in a 3U chassis. It provides 256 gateways, 8 PRIs and a switch that allows the insertion of a few ordinary MX-ONE circuit boards for connection of non IP terminals and networks.

The following interfaces and users are inherently supported by this media gateway:

- Redundant Ethernets. Either port can support both call control and media.
- 8 x PRIs (ISDN E1/T1)
- Mobile extensions
- IP extensions (H.323)
- IP extensions (SIP, including IP DECT and Wi-Fi)
- IP trunk (H.323, SIP)
- IP networking with other PBXs
- QSIG networking with other PBXs.
- InAttend for CMG, PC based SIP attendant
- BluStar clients (8000i and BluStar for PC, iPad and iPhone)
- 2 or 4 available board positions (for TDM interfaces) (1 if the ASU-E is used and 3 if an external server is used).

6.1.3.2

MX-ONE Classic

The MX-ONE Classic media gateway is provided as one 7U high, 19-inch wide subrack with a number of board positions for different functions and interfaces. The MX-ONE Classic is based on the MGU board.

The MX-ONE Classic can optionally be equipped with an ASU-E. The ASU-E is a standard PC embedded on a board hosted by the MX-ONE Classic.

This media gateway has the same interfaces as the MX-ONE Lite but provides more slots for added functions (boards).

6.1.3.3

MGU Board, in MX-ONE Lite and MX-ONE Classic

The MGU is a device board to be inserted in a dedicated board position in a 3U or 7U chassis/subrack. Unlike other device boards, this board is required in the Media Gateway subrack, and only one can be inserted at its dedicated board position. The key features of MGU includes:

- **Device Board Interface.** MGU intermediates all communication between device boards in the 3U/7U chassis and the Telephony Server.
- **Digital trunks.** MGU provides layer 1 and layer 2 for E1/T1. The MGU board has 8 Primary Rate ISDN (E1/T1) interfaces.
- **Keycode Receiver.** MGU provides Keycode Receivers (DTMF receivers), intended for mobile extensions.
- **Fax relay T.38.** MGU provides relaying T.30 facsimiles (G3 fax) to/from Internet Aware Faxes or Gateways using T.38 protocol.

- **TDM switch** for 512 time slots (replacing LSU-E and 2 DSUs). MGU provides a non-blocking TDM switch for inter-connection of circuit switched media.
- **256 gateway channels** (replacing 8 IPLUs)
- **Recorded Voice Announcements.** MGU provides play out of pre-recorded, locally stored, media files over TDM switch.
- **VoIP.** MGU provides RTP/SRTP including DTMF relay and facsimile tones over RTP.
- **Network Redundancy.** The MGU supports two types of redundancy.
 - Subnet (IP) redundancy
 - Switched (Ethernet) redundancy (Link Failover)
- For more information, please see the description for the *MGU*.

6.1.4 Earlier versions of Media Gateway Hardware

This chapter describes MX-ONE media gateways supported by MX-ONE 5.0 but not available for new deliveries.

6.1.4.1 Media Gateway Classic, equipped with LSU-E

MX-ONE Media Gateway Classic with LSU-E is provided as one or two 7 U high, 19-inch wide subracks with standing circuit boards.

6.1.4.2 Enterprise Media Gateway (EMG)

EMG is a self-contained, 19-inch, rack-mounted 1 U unit. It is not for new deliveries.

Note: This media gateway is not supported by the multiple gateways per server functionality. Each media gateway needs its own server.

6.1.4.3 Media Gateway with Legacy MD110 Hardware

A media gateway with legacy, pre-19 inch, MD110 hardware is any installed MD110 cabinet where the LIM switch board is replaced by an ethernet-connected LIM switch (LSU-E). Note that the MGU cannot be used in these cabinets. All pre-19 inch, hardware is service stopped and Astra give no guarantee for it's functionality.

For information about functions and interfaces, see the relevant MD110 documentation.

6.1.5 Protocols

For Voice Over IP (VoIP) communication, MX-ONE supports both IETF Session Initiation Protocol (SIP) and ITU-T H.323 open standards. MX-ONE provides full SIP extension functionality with the Aastra 6700i terminal models, and some functions also for SIP clients, like the BluStar client, and H.323 functionality in the MX-ONE Office IP phones. Basic SIP and H.323 functionality is available for other IP terminals.

The SIP extension functionality when used with the Aastra 6700i terminals is offering a higher level of end user services than the MX-ONE Office IP phones.

The SIP trunk solution in MX-ONE offers limited call functionality including caller ID/name DTMF digit signaling, diversion and bypass functions. The purpose of the SIP trunk solution is to enable communication with SIP based access to public networks, to the ACS attendant and to connect third party products, for example, video conference systems. For more information on which features that are supported by H.323 and SIP, see *MX-ONE FEATURE MATRIX*.

6.1.6 Server Options

MX-ONE offers a number of options for the physical server.

The common denominator for all server options is that the operating system is Novell SUSE® Linux Enterprise Server (SLES) version 10, Service Pack 4.

6.1.6.1 Standard Server

The turn key servers delivered with MX-ONE is 1U rack-mounted standard pre-configured server, with the following configuration:

HP Proliant 360 G7 Server:

- Intel Xeon E5620 processor (2.4 GHz Quad Core processor)
- 6 GB RAM (DDR3)
- 300 GB SAS HDD
- 4 LAN ports (100 or 1000 Mb/s)
- Internal DVD reader

DELL Poweredge R310 Server:

- Intel Xeon X3450 processor (2.66 GHz Quad Core processor)
- 8 GB RAM (DDR3)
- 300 GB SAS HDD
- 2 LAN ports (100 or 1000 Mb/s)
- Internal DVD reader

The standard server (or the ASU-E server board, see below) should be used to serve multiple media gateways.

6.1.6.2 **ASU and MX-ONE Server**

ASU Server consist of an ASU-E mounted in a 1U high unit in the 19-inch building practice.

The unit can be used as a stand alone telephony server for media gateways. This server option has very low power consumption. Being a standard server with proprietary form factor, it can also be used for any application on, for example, Windows.

MX-ONE server consists of an ESU built into a 1U high unit in the 19-inch building practice.

6.1.6.3 **Server for Stackable**

In the Stackable building practice ASU-E must not be inserted in the backplane. An external server solution must be used.

6.1.6.4 **Server for MX-ONE Classic/Media Gateway Classic**

For MX-ONE Classic/Media Gateway Classic, the server used is either an ESU or the more powerful ASU-E. The ASU-E is embedded on a single board hosted by the Media Gateway Classic and runs SLES 10 SP3.

ESU

The ESU is a standard PC embedded on a board hosted by Media Gateway Classic, with a 40 GB hard disk, a 1.4 GHz processor, and 2GB RAM. The operating system is Novell SUSE® Linux Enterprise Server (SLES) version 10, Service Pack 4. The ESU has the same function as the standard server, but with limited capacity.

ASU/ASU-E

ASU is a ASU-E mounted in a 1U high chassis.

ASU/ASU-E is a server board supported with MX-ONE 5.0. The ASU-E server board is based on standard COMXpress Modules and comprises the following characteristics:

- SATA 2,5" HDD (the HDD can be changed to SSD, if this option is preferred for better performance).
- 2 Ethernet ports
- 1 VGA port
- 4 USB 2.0 ports

Power consumption: 25-45 W.

ASU supports -48 V, 90 V to 240 V.

ASU-E supports -48 V.

6.1.7 Software Only Option

6.1.7.1 Non-virtualized environment

Telephony Servers can also be delivered as software only together with media gateways. For this type of packaging, the functionality of the operating system including Linux support subscription and used server is the responsibility of the distributor. Minimum requirements for the server depend on the number of users per server:

With up to 2 500 users per server:

- Processor 2 GHz Core 2 Duo
- RAM memory 4 GB
- Disk 72 GB
- Intel x86 architecture
- Chipset with watchdog implementation
- LAN ports: 2 (100/1000 Mb/s)
- USB: 2 (USB 2.0)
- DVD: internal or USB
- Console I/O

With up to 15 000 users per server:

- Processor 2.4 GHz Quad Core
- RAM memory 6 GB
- Disk 72 GB
- Intel x86 architecture
- Chipset with watchdog implementation
- LAN ports: 2 (100/1000 Mb/s)
- USB: 2 (USB 2.0)
- DVD: internal or USB
- Console I/O

6.1.7.2 Virtualized environment

For virtualized environment it is always recommended to follow VMware recommendations for host server configurations, but for running MX-ONE in a virtualized environment, the minimum configuration for the consolidated setup is the following:

- Host machine hardware minimum for consolidation for 1 MX-ONE Telephony Server with maximum 1000 users and 2 Calls/second using 0,2 Erlang as base.
 - Minimum 1 x CPU @ 2,4Ghz Intel Xeon with 6 cores supporting Hyper-Threading
 - Minimum 6 GB RAM
 - Server certified to operate with VMware
 - ESXi Hypervisor, minimum 2 GB RAM.
 - Telephony Server Guest machine, minimum 2 vCPU 1500 Mhz, 4 GB RAM and 60 GB HD for the Telephony Server 1.

Additional Telephony Server Guest machines, add minimum 2 vCPU 1500 Mhz, 4 GB RAM and 60 GB HD per extra Telephony Server guest machine. Note that it is in addition to the 6 GB above.

Note that the CPU above has 6 physical cores, so the maximum recommended numbers of Telephony Servers are 2, when the Manager Provisioning is used in the same physical host (another Virtual Machine).

For more information about MX-ONE in a virtualized environment, see *MX-ONE Telephony Server Virtualization Description*.

6.1.8 Combined Media Gateway and Server

MX-ONE Telephony System can be configured with any combination of servers and media gateways. The following pre-packaged alternatives are supported:

6.1.8.1 MX-ONE Lite with ASU-E

This MX-ONE server and media gateway combination consists of an MX-ONE Lite equipped with an MGU, an ASU-E, and a TMU board. This unit has two available board position for any type of existing interface (for example, DECT, ATS, DTS).

The MX-ONE Lite with an embedded ASU-E replaces the Compact SM for new sales.

6.1.8.2 Compact SM

The Compact SM is a 19-inch 2U high unit that includes an ESU board and an EMG. This unit is supported for existing installations upgrading to MX-ONE 5.0, it is no longer available for new deliveries.

Note: This package is not supported by the multiple gateways per server functionality (each EMG needs its own server).

6.1.9 Boards and Functions in MX-ONE Media Gateways

This chapter gives a general description of functions provided by the following MX-ONE media gateways:

- MX-ONE Lite (available in MX-ONE 5.0)
- MX-ONE Classic (available in MX-ONE 5.0)
- Media Gateway Classic equipped with LSU-E (not for new deliveries)

This chapter is not valid for EMG.

6.1.9.1 VoIP Traffic

The media gateways convert the RTP streams to TDM data, and TDM data to RTP streams. The conversion is used for gateway calls where the media streams are handled by a media gateway. For the RTP packet, G.711, G729A, G729B or G.729AB speech codecs are supported with 30 ms packet lengths. The speech codecs can be changed in steps of 10 ms. Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) are supported and can be enabled per call. Adaptive jitter buffer is also supported for a maximum of 240 ms buffer length. The buffer is configurable and allocated on a per call basis.

6.1.9.2 Echo Cancellation

MX-ONE media gateways supports echo cancellation complying with ITU-T G.168 standard. The echo canceller is used for the VoIP traffic. The echo cancellation algorithm manages up to 128 ms tail length.

6.1.9.3 Analog Interfaces

The ELU34 provides 32 extension lines for analog extensions. The extension lines carry speech signals, DTMF signals, and voice-band analog functions. The ELU34 board also provides DC feeding, loop detection, ground key detection, ringing trip detection, ringing voltage generation, and call metering functions. The ELU34 can be adapted for different market requirements and for ringing signals superimposed on 0 V or -48 V. The board consists of the analog extension line interfaces, two DC/DC converters, one ringing voltage generator, and one control part. The analog interface supports Calling Line Identification Presentation (CLIP and CLIR), which means that the phone number, for incoming calls, is presented on the display of the called party's terminal. The ELU34 uses two signaling methods to send out this type of information, either DTMF tones or Frequency Shift Keying (FSK).

6.1.9.4 Analog Trunk Units

Analog trunk units are provided with the TLU80 and TLU83 boards. The TLU80 board provides analog E&M and inband tone signaling. The TLU83 board provides analog loop and ground start signaling.

6.1.9.5 Digital Interfaces

The ELU33 provides 32 lines for digital extensions. It occupies 32 timeslots of the switch. The ELU33 can also handle enhanced signaling on the extension line. Firmware download to both the board and to terminals connected to the extension lines is supported.

6.1.9.6 Digital Trunk Units

The digital trunk units are mainly intended for connection with PSTN and PLMN networks but can also be used for private networking. The digital trunks can be used for ISDN/QSIG, DPNSS, and CAS signaling. The TLU76 board provides a single E1 (30B+D) interface, that can be used for ISDN/QSIG, DPNSS, and CAS signaling. The TLU77 board provides a single T1 (23B+D) interface, that can be used for ISDN, DPNSS, and CAS signaling. The TLU79 board provides four ISDN-T Basic rate interfaces (2B+D). Media gateways with the MGU board have 8 built-in E1/T1 ISDN PRIs.

6.1.9.7 Tone Receiver/Sender and Conference Functions

The purpose of the tone receiver or sender function is to send out and receive tone messages, such as dial tone, busy tone, congestion tone and DTMF tones. The function is required for extensions, trunk calls and conference calls. The TMU board is a mandatory board in all media gateways except EMG.

The TMU provides the function of tone/DTMF receiver, tone/DTMF sender, and conference functions. The TMU can be initiated to handle A-law or mu-law PCM. The SPU4 board is a board that provides the tone receiver function, used for DTMF detection from mobile extensions.

MX-ONE Classic, equipped with LSU-E, which shall handle mobile extension traffic, requires SPU4 boards, since all mobile extension calls will request a DTMF receiver in SPU4 (instead of TMU). MX-ONE Lite and MX-ONE Classic, equipped with MGU, have the DTMF receivers built-in on the MGU.

MX-ONE Lite is delivered with a 'built-in' TMU. This TMU does not have interfaces for external Music on Hold. To get Music on Hold in MX-ONE Lite, functionality in MGU must be used, or additional TMU must be inserted.

6.1.9.8 VSU

The VSU, Voice Storage Unit, is required only when Recorded Announcements are needed in a Media Gateway Classic equipped with LSU-E, and there is no MGU in the system, or when the MGU(s) cannot handle the needed RVA capacity.

MGU and EMG have built-in Recorded Announcement resources. A Media Gateway Classic with LSU-E can 'borrow' RVA resources from an MGU (but not from EMG).

6.1.9.9 MFU

The MFU is used for MFC signaling. Eight senders and eight receivers are handled by the board.

6.1.9.10 DC/DC Converter

The DC/DC converter is required in all media gateways except MX-ONE Lite and EMG. The DC/DC converter is a function on a board that provides the power supply, +/- 5 V; +/-12 V to all device boards through the backplane. One DC/DC board is used per subrack.

- Input voltage range: -47 to -58V uninterrupted

The DC/DC converter is also provided with an alarm input interface, which connects external alarms A and B to the backplane.

6.1.9.11 FTU

At power failure, the FTU board connects up to eight preselected analog phones directly to the public telephone exchange (PSTN terminals) through analog trunk lines. Normally the analog phones are connected to the ELU terminals, and the PSTN terminals to TLU terminals on the FTU2 board.

6.1.9.12 VCU

The Voice Compression Unit is used for the Dynamic Route Allocation feature. The VCU application on the SPU4 can handle two timeslots with four compressed channels each. Each channel can either be decompressed or switched to another timeslot. The VCU is not available for new sales.

6.1.9.13 LSU-E

The LIM Switch is a distributed function that handles the TDM switching function through the back-plane. It consists of five boards, one LSU-E,

and four DSUs, for a 1024 x 1024 timeslot switch. The LSU-E communicates with the telephony applications running on the server through an Ethernet interface. Internally, the LIM Switch consists of a control memory and a speech memory. The control memory stores the connection information while the speech memory stores speech samples.

The LIM Switch is also able to attenuate or amplify the audio sample that is switched through. Sixteen alternative levels are accessible in the LSU-E or DSU for A-law coding and there are also a number of defined levels for mu-law coding. The different alternative levels are market dependent. For more information, see the appropriate market characteristics documentation. The LSU-E is also equipped with a clock unit that has hold-over capabilities. Hold-over means that the external synchronization frequency can be memorized and synchronization maintained even if the external synchronization is lost for a short while. The LSU-E and DSU take synchronization from one selected device board. Each of the DSUs serves 256 timeslots and performs serial and parallel conversion of the Pulse Code Modulation (PCM) signals to and from the device boards.

This board is only used for migration of Stackable systems.

6.1.9.14

IPLU

The IPLU board handles rtp-resources, and media connections between or to the IPLU and a terminal (SIP or H.323). It supports up to 32 concurrent media channels. The IPLU has two external interface connectors on the board front, one Ethernet port and one RS232-port for debugging purpose. The device processor on the IPLU is equipped with 128 MB RAM for program and data memory and a 512 KB FLASH PROM for the boot program. The application program is loaded from the on-board Compact Flash card. The application program can be downloaded from a web server to the board. The IPLU supports T.38 fax for inter-LIM connections and for media connections to SIP end points. The IPLU also supports clear channel transport for inter-LIM connections over IP for, for example, basic rate modems connected to ISDN S0 extensions.

6.1.10

Group Switch

The group switch establishes the media connection between single server and media gateway units. The group switch is built up by Group Switch Modules (GSM). Each GSM can have 31 PCM lines connected to it. A group switch can have up to 8 GSMS, and it can be duplicated. For more information, see the operational directions for *GROUP SWITCH*.

The GJU-L board is used to connect a single media gateway to the Group Switch.

The Group Switch is only included in migrated or upgraded systems.

6.2 MX-ONE Manager Suite

MX-ONE Manager comprises the following management tools:

- MX-ONE Manager Telephony System (system management)
- MX-ONE Manager Provisioning (user and extension management)
- MX-ONE Manager System Performance (traffic measurement information)
- MX-ONE Manager Availability (fault and performance management)

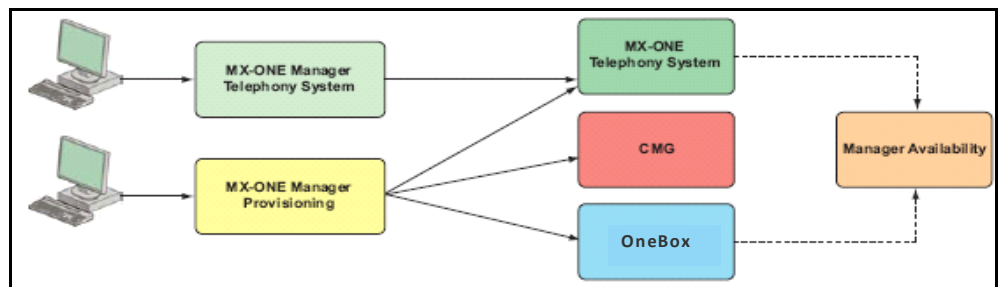


Figure 13:MX-ONE Manager

The functions in MX-ONE Manager are in accordance with the Fault, Configuration, Accounting, Performance, and Security Management (FCAPS) paradigm.

For more information, see the description for *MX-ONE MANAGER APPLICATIONS*,

6.2.1 MX-ONE Manager Telephony System

MX-ONE Manager Telephony System is a web-based application, accessed using a web browser. The application provides functions for configuring and managing MX-ONE Telephony System including, for example:

- Performing full setup of MX-ONE
- Managing media gateways
- Managing routes
- Managing attendants
- Managing groups, number plans, common categories, and service profiles
- Creating and maintaining configuration files for IP phones
- Monitoring IP phones

- Performing support tasks such as backing up and restoring data in MX-ONE Telephony System, viewing information about hardware and software revisions, and security and event logs.

The application comprises an integrated command line interface that can be used for entering commands manually.

6.2.2 MX-ONE Manager Provisioning

MX-ONE Manager Provisioning is the user and extension management application in MX-ONE, providing a single point of entry for managing user and extension data in MX-ONE Telephony System, OneBox, CMG, and AMC Provisioning Server.

MX-ONE Manager Provisioning also provides functionality for (for example):

- Managing administrator accounts
- Adding subsystems, for example, MX-ONE Telephony Servers and CMG servers
- Importing and exporting user and extension data
- Performing backup of user and extension data
- User account administration, for example, unlocking users.
- End-user self-service.

When changing user and extension data in MX-ONE Manager Provisioning the corresponding data in the MX-ONE Telephony System, OneBox, and CMG databases is automatically updated accordingly.

Note: The MX-ONE Manager Provisioning database is the master user and extension database in MX-ONE. MX-ONE Manager Provisioning must therefore be used when, for example, adding or deleting users. Changing user or extension data in CMG or MX-ONE Telephony System will cause unsynchronized data in the MX-ONE databases.

Application specific user and extension data, for example, time zone settings in CMG, is managed using the management tool of the specific application. Time zone settings, for example, are managed using CMG's *OfficeWeb* or *Directory Manager*.

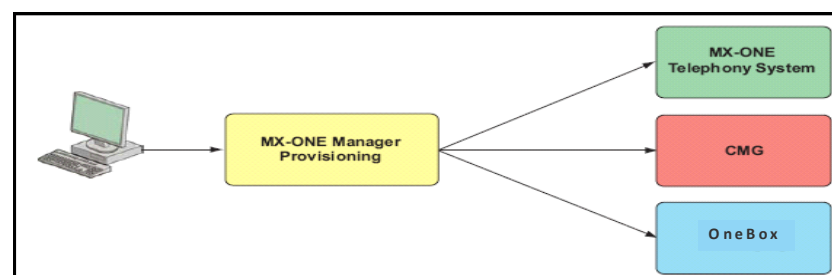


Figure 14: User and extension data flow in MX-ONE

For a detailed description regarding data management using MX-ONE Manager Provisioning, see 10.3 Configuration Management.

6.2.3 **MX-ONE Manager System Performance**

Manager System Performance (MSP) provides simplified measurement and analysis of performance data from MX-ONE. It also gives the MX-ONE administrator information about the overall performance of trunks, routes, operators, individual extensions and common system resources.

- Graphical interface for managing and viewing the MX-ONE performance.
- Easy to read and customize traffic information from the MX-ONE.
- Traffic analysis of major MX-ONE components, including Radio Base Stations and IP extensions.
- Multiple MX-ONE nodes can be managed.

MSP supports extended reporting or radio base station performance statistics. Performance data can be easily accessed from the MX-ONE through the simple user interface. This data is automatically retrieved and stored in the MSP server.

Using the Presentation Manager sub-modules, the data can be easily viewed, modified and presented in either report or graphic formats for printing or distribution within the organization.

With MSP, potential bottlenecks in the MX-ONE implementation can be avoided and system resources tuned to ensure maximum availability.

6.2.4 **MX-ONE Manager Availability**

MX-ONE Manager Availability provides fault and performance monitoring for MX-ONE servers and applications. Filtering and basic correlation is provided to ensure that the administrator/attendant can focus on important alarms and protect customers' management framework from overloads. Some of the key features included are:

- Server Monitoring - Monitors the performance, resource consumption and capacity of critical server components such as CPUs, memory, caches, physical and logical disks.
- Application Monitoring - Picks up and handles events, monitors status of critical applications.
- VoIP QoS Monitoring - Provides QoS information, for example latency, jitter, and packet loss. There is also an interface to provide information to an external system for more advanced calculations and presentation.

- Performance monitoring - traffic data and vital parameters are monitored.

Manager Availability is based on BMC ProactiveNet Performance Management with applications developed especially for MX-ONE communication solution. This application is a web-based management interface and the security communication is provided between managed system and Manager Availability via SSL.

6.3 Unified Communication Application Suite

6.3.1 OneBox

OneBox, also know as MX-ONE Messaging consists of features for voice mail and fax mail. OneBox provides capabilities to store and retrieve voice mail, e-mail, and fax.

OneBox also supports an interface allowing client applications, such as the Manager Provisioning, to create voice mailboxes on OneBox.

The operating system for OneBox is Windows Server 2003 R2. Windows 2008 Server support is planned for a later OneBox service pack.

For detailed information about OneBox, see the different documents (about Fax Mail and Voice Mail) in the MX-ONE CPI documentation under Messaging.

6.3.2 CMG Application Suite

The following CMG applications and products are available for MX-ONE:

- CMG Server, including Directory Manager (web-based), Configuration Manager (web-based), Phone Book (tool for formatting and printing). Note that Directory Manager is used for managing CMG specific user data only. For detailed information regarding user data management, see 10.3 Configuration Management.
- InAttend and NOW, SIP-enabled PC-based attendant solutions.
- Office Web (presence management, directory search) including Quick (taskbar program to quickly access presence information) and Web services (access CMG data using XML).
- Calendar Connection (server-based calendar integration for Microsoft Exchange, Lotus Domino, and Novell Groupwise)
- Visit (pre-registration, visitor self check-in, advanced booking, group booking, notification of arrival (e-mail or SMS), and badge printing)

- BluStar for PC Client (with easy access to a set of UCC features from a single interface, the user experiences an intuitive tool that unifies voice communications with video, instant messaging, directory look-up, flexible search options and conferencing.

For more information on the CMG application suite, refer to the CMG documentation.

6.3.3 Aastra Solidus eCare

Aastra's Solidus eCare is an advanced multimedia contact center offering agent, management, and self-service applications. It operates with MX-ONE and presents full IP-based contact center capabilities.

Solidus eCare communicates with MX-ONE using Aastra's Open Application Server (OAS). OAS is an open, scalable and distributed platform on which applications using Computer Telephony Integration (CTI) can be based. OAS call control model is based on the Computer-Supported Telecommunications Applications (CSTA) protocol model.

Solidus eCare is an all-in-one contact center that provides a solution for all types of businesses. Built as one fully integrated solution, Solidus eCare provides all your contact center needs such as multi media support, IP enabled, mobility for your agents, virtual contact center for your distributed sites and also tenanting when supporting multiple contact centers from one installation.

Solidus eCare is a multimedia contact center solution that provides end-to-end services to enterprise customers of the enterprise. It blends telephone voice calls, e-mails, SMS, internet collaboration, fax and, Web chat, and e-mails in a single virtual queue. Solidus eCare's intelligent skills-based routing capability ensures that customers are always routed to the best qualified agent.

Solidus eCare is a flexible, scalable, open, high availability solution. Solidus eCare can now evolve and grow with your business needs. Its open architecture allows Solidus eCare to integrate with customer business processes and applications such as work force management and customer relationship management applications.

For further information, please see Solidus eCare specific documentation.

6.4 Terminals and User Applications

The entry point for the user of MX-ONE is through the different clients. These clients can be either phones or software clients running as PC applications. The different terminals and user equipment use the telephony system for their Telephony Servers. For more information about

the telephony functions for the different terminals and user equipment, see *FEATURE LIST* and *CAPACITIES*.

The following terminal types and applications are supported:

- IP phone (H.323 or SIP)
- Analog phone
- Digital phone
- Mobile phone
- Cordless phone (DECT)
- Aastra SIP DECT phone
- IP DECT phone (Ascom)
- BluStar Aastra 8000i video terminal
- Wi-Fi phone
- CAS extension interface
- ISDN S0 terminal interface
- Paging equipment
- DRG, Digital Residential Gateway (analog-to-VoIP)
- Old terminals or clients (such as OPI-II)
- Aastra Mobile Client (AMC)
- BluStar for PC UCC client
- PC-based attendant (CMG InAttend and NOW)
- Office Web (CMG)

For more information regarding the terminal types, see the appropriate directions for use.

6.4.1

IP Phones

In MX-ONE Telephony System, the IP phones can communicate using either the SIP or the H.323 protocol. Note that it is not recommended to use H.323 and SIP phones simultaneously in the same MX-ONE system. This is because calls between SIP and H.323 terminals will be forced gateway calls, which will lead to low capacity. Any SIP- or H.323-compatible IP terminal (phone or client) may be connected to MX-ONE Telephony System. However, for full functionality, the Aastra IP phones or soft clients must be used.

The SIP extension solution in MX-ONE offers basic calls according to standards, including caller ID/name, but also services like hold, transfer and conference. Other services like call pickup, intrusion, call back are also supported, and initiated either with suffix dialing or with softkeys, depending on model.

Using the Aastra 6700i range of SIP telephones, the end-user feature level on high-end versions is similar to that offered by the Dialog series

using H.323, plus some more functions, like multiple terminal login on the same number (forking), Extra Directory Numbers, Shared Call Appearance, Intercom, Do Not Disturb and Call Parking pool.

Basic functions are available according to the H.323 standard, including Inherent Free Seating, allowing any IP telephony user to log on to any IP phone. To be able to offer voice routing services, like Inquiry and Transfer, the routines for rerouting of media according to ITU-T H.323 are used. The IP phones shall be supplied from Aastra to ensure the highest level of featured functions but can be of other brands, supporting the open H.323 or SIP standard.

MX-ONE supports IP phones from the Aastra 6700i series and the DBC 42x series and all newer terminals. Examples of features for the MX-ONE IP phones are:

- An automatic installation of configuration information and updated software from a Web server whenever the phone is restarted.
- A web browser interface enables users to make their own phone settings, such as edit a phone book, or a function key.
- Internet access.
- Monitoring keys.
- Importing contacts from Microsoft Outlook.

For detailed information on the IP phones, see the directions for use for each phone.

6.4.2 Analog Phones

Analog phones, answering machine systems, or Group 3 Fax with DTMF signaling capabilities can be connected to MX-ONE Telephony System. Rotary dialing phones only support basic call, they cannot use services.

For detailed information on analog phones, see directions for use for *ANALOG EXTENSION FOR MX-ONE*.

6.4.3 Digital Phones

An extension equipped with a digital phone can use the telephony features without dialing procedures, since the digital phones are equipped with programmed keys for the most-used features and programmable keys for other features (the most advanced features are soft-keys in combination with the display).

The following digital phones can be delivered with MX-ONE Telephony System:

- DBC 220
- DBC 222

- DBC 223
- DBC 225

In addition a number of older versions of digital phones are supported, see the description for *CAPACITIES*.

For detailed information on the digital phones, see the directions for use for each phone.

6.4.4 Mobile Phones

The Mobile Extension (ME) feature makes it possible to use mobile phones as extensions in MX-ONE Telephony System.

The mobile phone has access to functions and features comparable to an internal extension, such as callback and conference. Internal parties who call the mobile extension maintain full functions on such features as callback and camp on from operator. Additional phones can serve as alternate answering locations using a specific dialing procedure.

To map and route calls between the PLMN subscription and the extension in the PBX, it is recommended to have an agreement with a local mobile phone operator. No extended software or functions is needed on the mobile phone itself.

If an agreement with a local mobile phone operator is not available, the recommended configuration is to use the Aastra Mobile Client (AMC). If not using AMC, a user may call a predefined number in the PBX from any mobile phone and receive a dial tone. The user can either manually dial, using a phone book or calling card service after validation, or enter a valid PIN code.

For detailed information on mobile phones, see directions for use for *MOBILE EXTENSION FOR MX-ONE*.

6.4.5 Cordless Phones (DECT)

Using cordless phones enables users to make and accept calls at any location in the coverage area of its base stations. This feature is compliant to the DECT GAP/CAP standards, which ensures desktop phone speech quality and full security from wiretapping.

The cordless feature consists of a number of software units, specific hardware/firmware units, external Radio Fixed Parts (RFP) and Portable Parts (PP). Cordless extension is a fully integrated type in MX-ONE, and can use most of the features available in the system.

The cordless phones can only be used in an MX-ONE Telephony System including at least one MX-ONE Classic or Media Gateway Classic.

For more information on cordless phones, see the description for *CORDLESS PHONE*.

The Short Message Service (SMS) performs transfer of text messages, which can be up to 140 bytes long (giving, for example, 160 Latin characters). SMS is available for Cordless extensions (DECT) in MX-ONE Telephony System. Text messages can be received in any call state, that is, for example, during an ongoing call. To be able to send and receive SMS messages, a license must exist for the extension on an individual basis.

For more information, see the operational directions for *SHORT MESSAGE SERVICE, MS*.

6.4.6 IP DECT System

The IP DECT system supports the DECT standard which gives a full integration of messaging and voice functions. The DECT system can be integrated with external applications such as different alarm systems, networks and e-mail. This gives features such as; messages to handset, alarm from handset, message acknowledgement, and absent handling.

For more information on cordless phones, see the description for *IP DECT SYSTEM*.

6.4.7 Aastra SIP DECT System

As an alternative to the Integrated DECT system, MX-ONE has full support for the Aastra OMM SIP DECT system, which includes the Aastra 6x0d Cordless terminal portfolio and provides full integration with OMM alarm and messaging system. The main difference is that RFP L3x SIP DECT base stations are connected to the IP network and the Aastra 6x0d DECT terminal registers to MX-ONE as a full-featured SIP extension

The Open Mobility Manager (OMM) supports features such as; update of handsets over the air, messages to/from handset(s), alarms from handset, man-down support (terminal dependent), terminal location services and message acknowledgement handling. Additionally, the OMM system can be integrated with external applications such as different alarm systems, Messaging applications, networks and e-mail using standards based protocols, such as XML or POP. As the DECT handsets register as SIP extensions, the necessary SIP user licenses need to be present in the system.

For more information on Aastra OMM SIP DECT cordless solutions, see the description for *SIP DECT SYSTEM*.

6.4.8 CAS Extension Interface

The CAS extension interface provides a digital connection to external equipment and provides, through PCM-links, the analog extension functions.

Using the CAS extension feature, different signaling systems can be handled simultaneously, which allows connections of different external equipment, for example, cordless phone system, digital multiplexers, voice mail, to an MX-ONE system.

6.4.9 ISDN S0 Terminal Interface

The ISDN S0 terminal interface, also called a Basic Rate Access (2B+D), is an interface to which ISDN terminals can be connected. An ISDN terminal interface has two 64 kbits/s B-channels, and two equipment positions are needed for one interface. The ISDN terminal interface allows voice and data transmission on ordinary phone lines.

One or several ISDN terminal directory numbers are affiliated to an ISDN S0 terminal interface. An ISDN terminal directory number is not affiliated to a certain ISDN S0 terminal, but it gives an access with some specific characteristics to or from that ISDN S0 terminal interface. There is no support for busy state services.

6.4.10 Paging Equipment

Paging enables persons out of hearing range of their extension instrument to be reached from an extension, PBX operator console or external line. The paged person is alerted of incoming calls by activating a portable personable radio receiver (staff locator) or signaling with a personal code over an acoustical/optical system. Answers to paging are mainly executed as meet-me.

There are three types of paging:

- Paging with meet-me answer only
- Paging with voice over radio and facilities for meet-me answering
- Paging with transmission of digital information to a display in the radio receiver and facilities for meet-me answering

6.4.11 Aastra Mobile Client (AMC)

The AMC application provides a graphical end-user interface for utilization of MX-ONE features. The AMC also provides call routing without mobile operator support and advanced cost saving features for internationally roaming users.

6.4.12 BluStar for PC client

BluStar for PC client delivers high-quality audio and HD video as well as access to a set of UCC features from a single client on the user's desktop. The direct integration with the MX-ONE Telephony System 5.0 provides the benefits of reduced cost and infrastructure complexity and secured high performance call functionality.

With easy access to a set of UCC features from a single interface, the user experiences an intuitive tool that unifies voice communications with video, instant messaging, directory look-up, flexible search options and conferencing.

6.4.13 InAttend

The InAttend is a scalable attendant console based on open standards. It is the core application in Aastra's attendant offering and an essential part of the Aastra UC suite. In addition to advanced call handling, InAttend offers powerful search options, calendar information, Microsoft OCS/Lync and IBM Lotus Sametime presence integration and all necessary information for efficient call handling. InAttend is also available for visually impaired attendants.

6.4.14 NOW Attendant

The NOW Attendant is a Windows®-based application for operators. A phone is necessary for providing media capabilities.

The operator may speak with either party individually or in conference. When the correct destination has been reached, the two parties are connected and the NOW is freed, ready to handle another call.

Additionally, the NOW is used to schedule user activities or other assistant functions.

The NOW is also available for visually impaired.

For detailed information on the NOW, see the user manual for *NOW*.

6.4.15 Old terminals or clients

6.4.15.1 Legacy Operator Instruments (OPIs) such as OPI-II

The DGF 220 10 is an operators instrument for MX-ONE. The product is not for new delivery, but can still be used with MX-ONE. Older OPIs such as OPI-II work for MX-ONE.

For more information, see the description for *PABX OPERATOR INSTRUMENT, OPI-II (2B+D) FOR ASB 501 04*.

6.5 High Availability

MX-ONE offers high availability by supporting the following types of redundancy in MX-ONE Telephony System:

- Network redundancy, where two types are currently supported:

- Dual subnet network redundancy

This type of network redundancy is achieved (provided a redundant network infrastructure is available) by connecting Telephony Servers to two LANs. If one of the LANs fails, the other one will continue to serve the operations and the Telephony Servers will be available on the functioning LAN.

Media gateways using MGU boards support network redundancy. These media gateways will function on the other network after the disturbance caused by the switch over at network failure.

In a Media Gateway Classic with LSU-E configuration, the capacity for gateway calls will be reduced. A switch must be used to secure control of the hardware when a LAN fails.

If EMGs are evenly distributed over the two LANs, approximately half of them will still function. During the downtime, the overall capacity of media processing will be reduced.

- Network redundancy by using Ethernet bonding in the Telephony Servers and Link Failover in the Media Gateway MGU.

Ethernet bonding in the Telephony Servers

By using Ethernet bonding, a Telephony Server is connected to two separate switches in a switched network. Provided that the switched network is designed for redundancy, this provides network redundancy to the Telephony Server.

Ethernet bonding means that two Ethernet interfaces are aggregated to work together. One interface is active at a time and the other interface is backup. The two interfaces share the same IP and MAC addresses. If one of the interfaces fail, the

other one will continue to serve the operations and the Telephony Servers will be available on the functioning interface.

Ethernet bonding is only supported in the Telephony Server Link Failover in the Media Gateway MGU

A proprietary active/backup (link failover) network redundancy technique is used with the MGU in MX-ONE 4.1 SP1 and later releases.

This means that the MGU's two Ethernet ports are each connected to a different switch, where one interface is active and the other interface is in passive mode. The two physical interfaces share the same IP and MAC addresses (for both media and signaling). If the active interface fails, the other one will become active and take over the traffic (both signalling and media) and the MGU will continue to operate normally.

Although the mechanism used by MGU differs from that of Telephony Server, they provide similar functionality.

LSU-E and EMG Gateways

Older Media gateways with only one Ethernet port (EMG or LSU-E) can be connected via a local switch. By using a local switch, which would be connected to 2 different switches on the same network, most capabilities in the network would be restored if there were network faults. The impact on system behavior depends on the time it takes for the network to recover. Depending on type of network disturbance with a Media Gateway Classic with LSU-E configuration, the capacity for gateway calls could be reduced. A local switch must be used to secure control of the hardware when a LAN switch fails.

- Server redundancy

Server redundancy is achieved by adding one or more standby servers to the network with the ability to take over any failing Telephony Server in the cluster. At failover, the standby server will take over the identity of the failing server and the control of the media gateways in the failing server. In case of a server failure ongoing direct media calls will remain connected but TDM and gateway calls will be disconnected. All new traffic will be redirected to the standby server.

In a distributed system connected over limited bandwidth (WAN), each remote domain must have its own standby server.

A prerequisite for server redundancy is that network redundancy is implemented to avoid any single point of failure.

- HLR redundancy

HLR redundancy (Home Location Register redundancy), or HLR backup, is a unique MX-ONE feature for IP extensions based in the same concept of HLR/VLR (Home Location Register / Visitor Location Register) from Mobile networks.

In a multi-server system it allows the possibility to register IP extensions with another Telephony Server (backup HLR), if the ordinary HLR server cannot be accessed. The feature is supported for H.323 and SIP extensions.

- Virtualization

A variant of Server redundancy and high availability can also be achieved by using the VMware Virtual Appliance functionality. See the description of MX-ONE Telephony server virtualization for details.

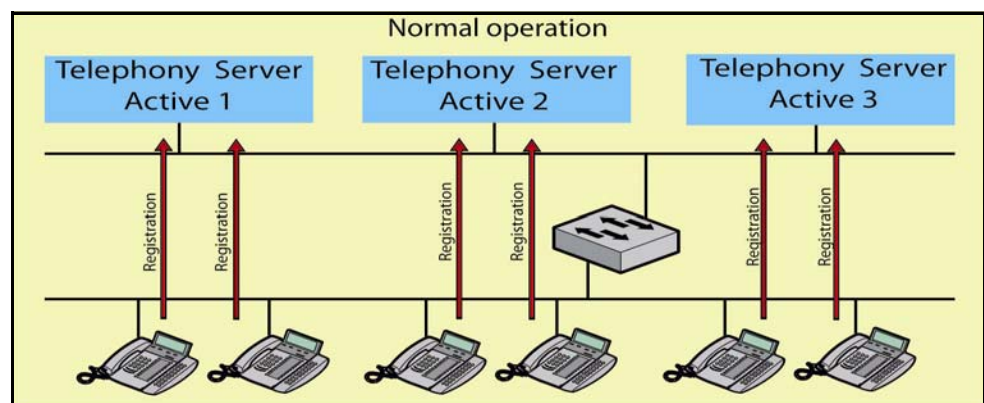


Figure 15: Normal operation

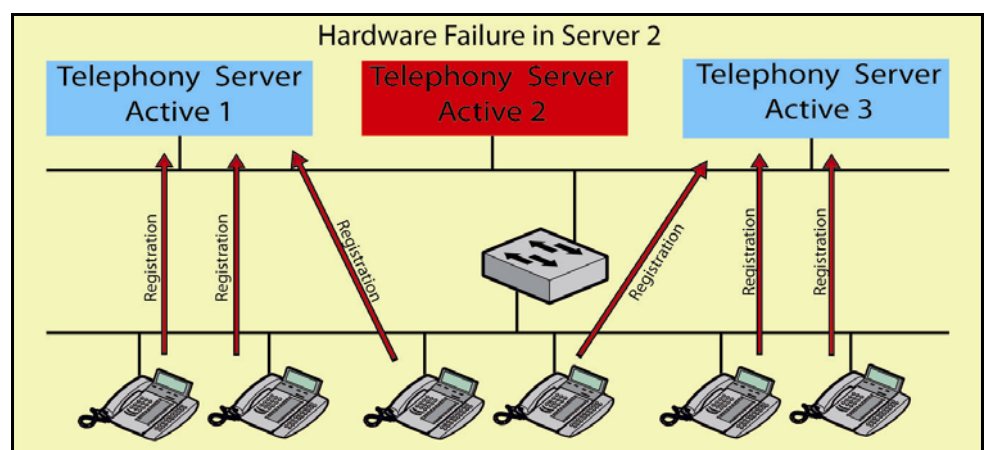


Figure 16: Hardware failure in Server 2

Note: From MX-ONE 4.0, the database structure has been improved by implementation of an internal LDAP database to store system data including HLR data. The data is now replicated across all servers, to insure no single point of failure if the user's "home

server" should go down. This enables users to be able to register on another (visitor) server and work as usual.

6.6 Scalability

With multiple media gateways, an MX-ONE standard server can handle up to 15.000 SIP or 5.000 Mobile Extensions. To scale upwards towards several hundred thousand users, up to 124 servers can be combined into a single system.

By using MX-ONE's IP networking or ISDN/Q-SIG+ networking (including VAPA services, Value Adding Proprietary Add-ons), multiple MX-ONE systems can be networked to create solutions for even larger geographically dispersed customers.

6.7 Networking Capabilities

Networking means to provide all services over network links to other entities of the Private Telephone Network (PTN). This cuts costs compared to the TDM-based leased line. Furthermore, it simplifies Branch Office scenarios to better fit the data network model with IP-connected Branch Offices.

To supply basic and supplementary services transparently, ISDN signaling over the Q-reference point (QSIG) is implemented over H.323 in IP networks. This makes networked PBXs possible over IP intranets and also over the Internet with the appropriate protection. The intra-connectivity over (private) ISDN is in accordance with ITU-T QSIG for E1/T1. Apart from QSIG, proprietary intra-connectivity is supported for the US market. MX-ONE supports both E1 and T1. The realization of the networking functionality in MX-ONE is implemented in the MX-ONE Telephony System.

Third-party products and VoIP service providers can also interoperate with MX-ONE using SIP signaling, for basic services such as basic call, caller ID or name, and DTMF digit signaling and supplementary generic services like parking, transfer, and conference.

For detailed information, see the descriptions for *NETWORKING* and *IP NETWORKING*.

7 Telephony Server Software

This chapter describes the MX-ONE Telephony Server software.

7.1 General

The same Telephony Server software is used on any server.

The software consists of the following parts:

- Linux operating system
- Service system
Software acting as an interface between the telephony applications and the operating system. For more information, see section 7.4 Service System.
- Telephony applications
- Manager Telephony System (section 6.2.1 MX-ONE Manager Telephony System)
- Manager Provisioning (section 6.2.2 MX-ONE Manager Provisioning)

7.2 Operating System

The operating system is Novell SUSE® Linux Enterprise Server (SLES) version 10, Service Pack 4.

7.3 Telephony Applications

Software that executes the call handling functions and has full control of all ongoing calls as well as resources in the media gateway.

For more information, see the description for *FEATURE LIST*.

7.4 Service System

The service system includes a number of functions used to support the telephony applications like service functions, system configuration change, functional change, start and restart of MX-ONE Telephony

System, data backup, license handling, system supervision, alarm handling, and automatic fault administration.

A program unit as described in this document is the telephony term equivalent to the computing term "process", which is a running instance of a program, including all variables and other states. From this point on, only the term "program unit" will be used.

A common function is a program unit that is centrally located and can be used by all program units in all servers. Regional functions are program units that exists in several or all servers, executing independent of each other.

For more information about task procedures, see operational directions for *ADMINISTRATOR USER'S GUIDE*.

7.4.1 Service Functions

The service functions is a collection of general-purpose functions and is used by the telephony application system and by the service system for the following:

- System status monitoring
- Server configuration
- Server status
- Program unit configuration
- Hardware configuration

7.4.1.1 System Status Monitoring

The system status monitoring function is used to monitor the system status at any given time to deal with special conditions, such as load, dump or restart.

System status monitoring includes the following actions:

- Prevent any change of reload data through commands during load, dump, or restart of any part of the system
- Prevent any change of reload data through programming from voice terminals during load or dump

The reload data changes are not prevented in the entire system at the same time, but in one server at a time during the action in this specific server.

7.4.1.2 Server/Gateway Configuration

The server configuration function is used to keep and provide status information regarding one server. It can also display information about

the total number of servers that exists in the system, including their server numbers.

7.4.1.3

Server Status

The server status function is used to keep and provide status information about all blocked or isolated servers.

- Blocked server

A server can be blocked manually from traffic or blocked by the system, if the server is faulty. Information about blocked servers is obtained by requesting status information for one or several servers. By using a subscription feature, the information is obtained every time a server is blocked or deblocked.

- Isolated server

When a server becomes isolated from the rest of the system due to, for example, a network failure, the application programs in the isolated server will obtain information about the current status.

7.4.1.4

Program Unit Configuration

Every program unit has a name and a unique program unit number and is also assigned a set of characteristics. This set of characteristics defines the program unit type, for example, such characteristics can be whether or not the program unit handles device boards, or if the program unit is issuing alarms.

The purpose of the program unit configuration function is to keep and provide information about the program unit types, including the assigned characteristics sets.

The following information can be obtained from the system:

- Configuration of common function - information about the servers in which a common function program unit is loaded
- Configuration within the server - information about whether or not a program unit is loaded in a selected server
- Program unit type - information about the selected program unit type

It is also possible to obtain information about the program units in which certain types or characteristics belong.

- Program unit identification - information about program unit number, name, and revision

7.4.1.5 Hardware Configuration

The hardware configuration function is used to give the following information relevant to device boards:

- Hardware position
- Board type
- Controlling program unit (if the board is initiated)
- Pointer to the controlling program unit
- Number of hardware individuals on the board (for example, number of supported extensions)
- Whether or not the board is blocked

There are two types of device boards: physical and virtual. The hardware configuration function is valid for both physical and virtual device boards. MX-ONE Classic and Lite have both physical and virtual device boards.

On request, non-initiated hardware individuals on device boards of a certain board type can be obtained, or the selected hardware individual can be translated to a hardware position by the system. The hardware position can, in the same way, be translated to a selected individual.

On request, general media gateway information, such as revision information, can be obtained. There is also a possibility to set or get information about a certain resource in the gateway. Examples of resources are virtual device boards, auxiliary devices, as well as the gateway itself.

7.4.2 System Configuration Change

The purpose of the system configuration change function is to adapt the system to the actual requirements of the customer and the environment.

It is possible to reconfigure the following objects in the system:

- Physical units
 - MX-ONE Telephony System — Telephony Server (for example adding a new server to an existing system)
 - MX-ONE Telephony System — Media Gateway Classic (for example adding a new board)
 - MX-ONE Telephony System — Media Gateway Lite
 - MX-ONE Telephony System — Media Gateway Classic
 - MX-ONE Telephony System — EMG
- Logical units
 - Program unit

7.4.3

Functional Change

The purpose of the functional change function is to carry out an update or upgrade, that is, to replace one or several program units in the system with new versions.

The function includes the following actions:

- Prepare the system for program unit change.

Commands changing the reload data are prohibited during the duration of the functional change.

- Add new program units either by specifying on the command line or using a file containing the program units. This can be done by the upgrade script, or it can be done manually when installing a hot fix.

The program unit can be assigned a group number to coordinate the change with other program units. The program unit can be a common or regional function. In the former case, the program unit is normally loaded in one server. In the latter case, the program unit is normally loaded in all servers in the system.

- Change to the new program units, which means that the system changes from the original program units to the new program units.

It is possible to group program units together to coordinate the program.

- End the program unit change.

Commands changing the reload data are prohibited during the duration of the functional change.

- Remove old program units from one or more servers by using a command.

This can be done by the upgrade script, or it can be done manually when installing a hot fix.

Program units that are absolutely essential for program execution in a server cannot be removed.

If a fault occurs that generates a data reload after removal of program unit or units and before a data backup is completed, an alarm is sent and the affected program unit's data is not reloaded.

- Display program unit change information, which comprises the following:
 - Added program units and their group affiliation
 - Revision information
 - Active and passive program units versions

- Process ID and path to the executable file for new and old program unit versions

7.4.4 Start and Restart of MX-ONE Telephony System

7.4.4.1 System Start

The purpose of the system start function is to set the system into operation after an update, upgrade or boot procedure.

System start comprises two main functions:

- Initial loading, which means that program units are loaded into memory.
- Initial start, which means that values are assigned to variables to activate the system functions and to link together the system logically.

7.4.4.2 Initial Load and Start

The initial load and start function comprises the following steps:

- Load all program units into memory.
- Assign values to variables of all program units.
- Start the program units.

7.4.4.3 Manual Restart

The purpose of the manual restart function is to recover manually from any kind of system fault.

The following can be manually restarted:

- The system
All program units are restarted. All traffic in the exchange is stopped and for example, IP phones are unregistered.
- Servers
All program units in the selected servers are restarted.
- Program units
Selected program units are restarted.
- Device boards
The restart is carried out by an activation request from the device-handling program to the stated board.
- Media Gateway

Selected media gateway is restarted and all gateway calls will be stopped in the server.

A manual restart is initiated by a command. It is not possible to change reload data, using commands during the restart phase.

7.4.5 Backup

7.4.5.1 Data Backup

The purpose of the data backup function is to backup and restore exchange data. Data backup is initiated by a command or done periodically. If the backup fails, an alarm is sent.

The following types of exchange data can be restored:

- System configuration data
- Application data (for example, extensions, and trunks)

The system can store five backup versions. If five versions already exist when a new backup is created, the oldest version is deleted.

The purpose of the safety backup function is to create a data backup stored on a different media, such as tape or another hard disk, or stored at a different location, for example, in a safe.

It is possible to specify the data that should be included in the safety backup and also what device to which the safety backup should be written. The system administrator decides how often a safety backup should be performed and how data should be stored.

If there is an inconsistency in the Telephony Server data, the safety backup can be restored to the master server. After having restored the master server, the configuration mirror can be distributed to all servers.

Exchange data of the entire system is restored from a backup. It is only the latest backup that can be restored. To restore an older version, a file has to be modified manually. After a data restore, the system is started. It is not possible to change exchange data during restoration.

7.4.6 License Handling

Server number 1 of a multi-LIM configuration always hosts the license server.

The purpose of the license server function is to prevent unauthorized use of system resources. The license server can be used in two ways, to check if a service is available, and to seize and release licenses.

The license file is delivered to a site in encrypted form and will be installed into the license server in encrypted mode.

The following two license types exist:

- Individual license

The individual license is used for ports, for example, the analog extension. Each individual license is defined with a maximum number of allowed licenses.

Any object that requires licensing seizes a license when the object is initiated. The license is released when the object is removed from the system.

- System license

The system license is used for optional system facilities, such as network services. A system license can have two values: service available or service not available.

The license information is stored and later checked when the optional system facility is initiated.

7.4.7 System Supervision

The system supervision function has the following purposes:

- Control the performance of the supervised objects, such as servers, gateways, boards, and alarms
- Fault detection
- Initiate fault correction measures
- Isolate faults

7.4.7.1 Supervisory Functions on System Level

Server/Gateway Supervision

The server/gateway supervision function has the following purposes:

- Supervise the start state and communication capabilities

This is done to make sure that all servers start up correctly and that inter-server communication is working.

- Build up, distribute, and maintain system tables

This includes the Common Function Table (CFT). The CFT specifies the server in which the different common function programs are running.

Cross Address Check

For some applications within the system, connections between data areas in different program units are built up. If the system consists of

several servers, it is most likely that these connections are set up between program units in different servers.

A distributed system solution implies that if a server fails, no traffic except traffic through this server will be disturbed. To avoid disturbances, no data lock up in other server should occur due to the faulty server.

If a software connection between two servers is released only at one end, data areas will remain locked as long as the linkage to the area is lost. This can lead to a congestion situation. To handle this lock up problem, each application program creating a mutual connection with data areas in other program units are instructed periodically to check the validity of assumed connections.

The check is administrated by the service system and is spread in time to avoid congestion.

Program Configuration

The purpose of the program configuration supervision function is to check that a common function unit only exists in two servers in the system.

Any conflict in the configuration causes an alarm.

System Backup

The purpose of the system backup supervision function is to check that a valid system backup is always accessible.

7.4.7.2

Supervisory Functions on Media Gateway Level

Test of Hardware before Loading

The purpose of the test of hardware before loading function is to validate that the hardware is functional.

All tests are done by the running operating system and BIOS functions.

LIM Switch Supervision

The LIM switch supervision function has the following purposes:

- Periodical testing of the switch board circuits using a supervision signal to LSU-E. The circuits are tested by internal firmware on the boards.
- Supervise connection setups
- Analyze device board supervision results

The LIM switch supervision is only applicable for a Media Gateway Classic configuration.

For more information about the LIM Switch, see section 5.1.9.14 LSU-E.

Supervision of Voltage

The purpose of the supervision of voltage function is to check the voltage of +5 V, -5 V, +12 V, -12 V, -48 V.

When a fault occurs, an alarm is raised containing information about the relevant voltage and the location of the missing voltage.

This function is only applicable for a Media Gateway Classic configuration.

Device Boards

The purpose of the device board supervision function is to test the device boards' communication abilities by sending a periodical test signal to the device processor. An immediate answer is expected.

The answer signal contains the identity and the number of individuals of the device board. The data is compared with the current board table to find any conflict in the device board configuration, that is, a newly inserted board can hide the signaling from a previous board, which will result in that the previous board is out of order.

This function is applicable for a Media Gateway Classic and MX-ONE Lite configurations.

Program Execution

The purpose of the program execution supervision function is to supervise the following:

- Execution times of a program unit

If a program unit is looping more than the allowed time due to programming faults or incorrect data, it will be interrupted and the job discarded.

- Address control

Situations can arise when the linkage between data areas has become faulty. To protect data areas from being accessed by other, unauthorized applications, an address control value is added to the pointer pointing out the data area. The address control value is a random value unique for the current linkage. The value is only valid as long as both areas agree in the linkage.

- Signal routing control

All signals exchanged between and within program units are checked in the following way before they are allowed to start execution in a program unit:

- Receiving server number corresponds to the server number address in the signal
- Addressed program unit exists

- Signal entry in addressed program unit exists for the sent signal
- Address control value correct

If any fault is found, the signal is not executed and the job is discarded. The signal that caused any of the above errors is stored for manual fault investigation.

Server/Gateway Disturbance Counter

Each LIM has a disturbance counter that reflects the quality of the program execution in the LIM. The counter is incremented a number of steps each time a fault occurs. The number of steps depends on the severity of the disturbance.

An alarm is sent when the first execution fault after start or restart occurs and when the counter exceeds the maximal allowed value. The counter is periodically decremented to avoid exceeding the value.

7.4.8 Alarm Handling

Alarm handling is handled by the service system. For more information about the alarm handling function, see 10.2 Fault Management.

7.4.9 Automatic Fault Administration

Automatic fault administration is handled by the service system. For more information about the automatic fault administration function, see 10.2.1.1 MX-ONE Telephony System.

7.4.10 Signal Tracing

The signal tracing function implements both the trace error log and the general trace function. The function implies that signals or text items are copied and preserved as the program execution is handling an activity so that the function can be analyzed later on.

The first trace individual is used by the error log function and is therefore always enabled in all servers.

Up to fifteen signal traces can be specified simultaneously.

The following four types of signal traces exist:

- Error log
 - Error log is always enabled and stores information about error situations in the software.
- Sequential tracing

Sequential tracing is used for tracing an activity. Through the sequential trace, the signals can be followed in their logical and chronological order and thus the chain of events that led to the fault. Sequential tracing should be used when the user has an idea of the activity that went wrong and wants to locate the break in the sequence.

- Interface tracing

Interface tracing is used to trace all signals to and from a program unit or a hardware equipment position. This form of tracing is used, for example, when the unit where the fault occurs is already known or suspected, but the process that caused the fault is not known (for example, when a faulty signal in a unit has been received).

- Hardware position tracing

Hardware tracing is used when the problem is related to a hardware board where only the signaling to and from the board is wanted. This function is especially useful when several program units are cooperating to the same hardware position.

8

Interfaces and Protocols

This section lists the external interfaces and protocols used in MX-ONE.

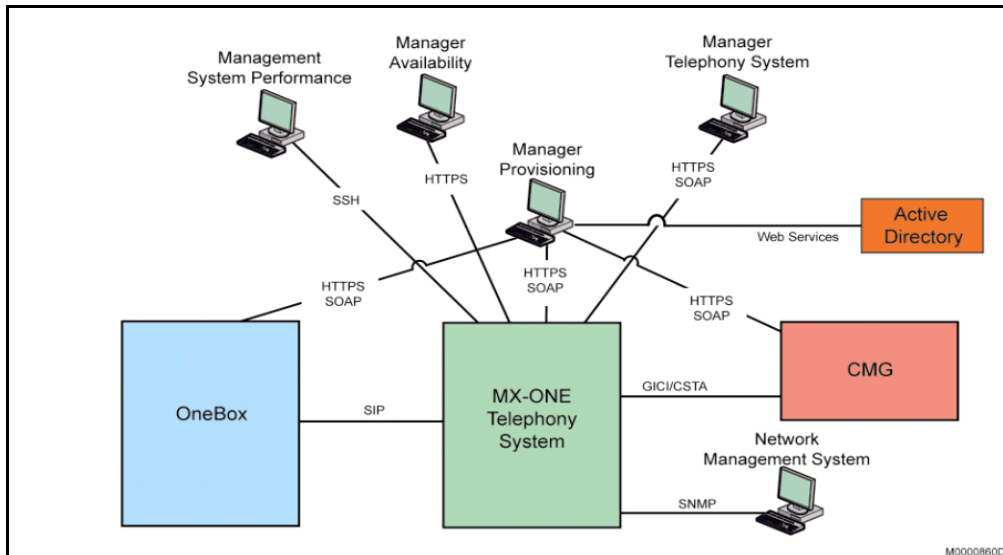


Figure 17: Interfaces and protocols in MX-ONE and MX-ONE Manager

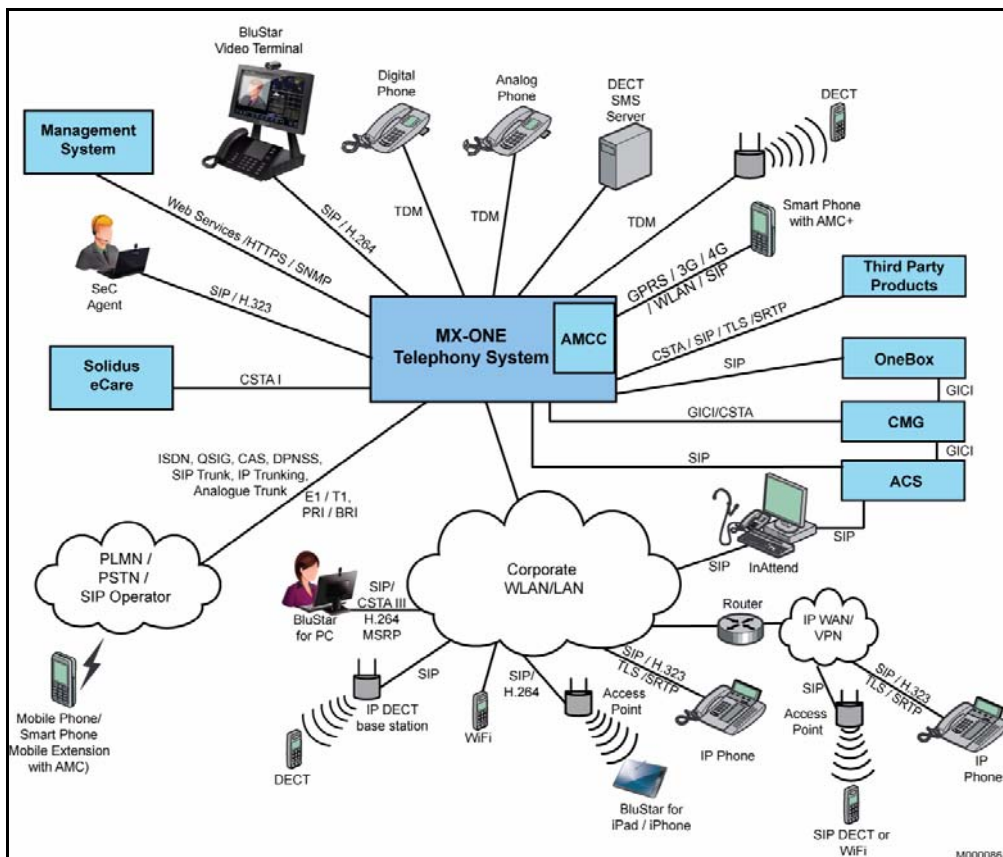


Figure 18: Interfaces and protocols in MX-ONE

The following interfaces and protocols are available for MX-ONE Telephony System:

- Private networking by means of ISDN QSIG over E1 (30B+D) and T1 (23B+D), the latter only in MGU.
- Private networking by means of ISDN tie line over E1 (30B+D) and T1 (23B+D) using proprietary signaling
- MFC
- CAS
- DASS2
- CCS7 (Simplified TUP for China)
- Private networking by means of DPNSS and CAS over E1 (30B+D) and T1 (23B+D)
- Public ISDN over E1 (30B+D), T1 (23B+D) and BRI (2B+D)
- H.323
- SIP (also used for IP DECT, SIP DECT and WiFi)
- SNMP
- Secure Shell (SSH)
- HTTP(S)
- Interfaces to an accounting system
- DECT
- DASL

The following interfaces and protocols are available for CTI:

- Telephone API (Microsoft)
- TSAPI (Novell)
- CSTA Phase III with XML and ASN.1
- TR87
- CSTA Phase 1 using Application Link

The following interfaces and protocols are available for CMG:

- GICI
- Attendant Interface
- LDIF
- Voicemail Systems Interface (VSI)
- HTTP/HTTPS for web applications
- HTTP Calendar Interface
- CMG Web services Interface
- Time-entry System Interface (TSI)

The following interfaces and protocols are available for OneBox:

- SIP

- H.323
- SNMP
- VSI
- Internet Message Access Protocol (IMAP) v4
- Messaging Applications Programming Interface (MAPI)

The following interfaces and protocols are available for Manager Provisioning:

- Web Services
- XML
- HTTP
- HTTPS

For detailed information about the different external interfaces and supported standards, see the description for *MX-ONE SYSTEM PLANNING*.

9 Migration

MX-ONE supports migration of MD110/Telephony Switch to an MX-ONE 5.0 system. The major steps of the migration procedure that uses Stackable or MX-ONE Classic magazines include:

- Install the servers needed after migration.
- Install version 5.0 and configure the system.
- Make a version 5.0 backup of each server.
- Stop the MD110/Telephony Switch and replace a limited number of boards.
- Connect the media gateways to the servers.
- Start MX-ONE 5.0

For detailed information, see installation instructions for *MIGRATING TSW/MD110 TO MX-ONE 5.0*.

10 Operation and Maintenance

10.1 General

Operation and maintenance in MX-ONE complies with the concept of Fault, Configuration, Accounting, Performance, and Security Management (FCAPS) for network management and consists of functions to supervise, administer, and manage the system.

This chapter describes operation and maintenance in MX-ONE from an FCAPS perspective.

10.2 Fault Management

MX-ONE provides two levels of Fault Management:

- Basic Fault Management (always included)
- Advanced Fault Management (using the optional product Manager Availability)

10.2.1 Basic Fault Management

10.2.1.1 MX-ONE Telephony System

Alarms

The *alarm* feature is used to store information on faults and disturbances in- or outside a system and to bring the attention of service personnel to these faults and disturbances.

Alarm classes

Each alarm is assigned one of the following alarm classes:

Severity	Description
4	Critical
3	Alert
2	Warning
1	Information
0	Cleared

The alarm class 0 (cleared) is special. When the alarm sender notices that an alarm condition no longer exists, the sender will clear the alarm and the alarm class changes to 0.

Alarm Generation

The alarm classes (except alarm class 4) are assigned a limit for incremental alarm. This means that an incremental alarm is generated when a predetermined number of alarms with a certain alarm class have been stored in the alarm log. The incremental alarm is then assigned the next higher alarm class. The limits for incremental alarms are specified in a configuration file.

An alarm is also generated when the alarm log risks becoming full.

Alarm Actions

Alarm actions can be configured to run when an alarm is stored in the alarm log, or when an alarm changes severity, for example, is cleared.

The alarm action can be any program or script running on Linux. Examples of actions that can be taken are:

- Sending e-mail or SMS with alarm information
- Updating web pages.

Alarm Presentation

Alarms can be presented in the following ways:

- As an alarm log printout (can be in any of eight available formats)
- As continuous alarm log view
- As e-mail or SMS
- On a network management center/workstation

Alarm Clearing

When the system detects that an alarm condition (fault) no longer exists, the alarm log is informed and the alarm is cleared.

A cleared alarm has the alarm class 0 but information on the old class is kept.

Alarm Noticed

When an alarm is noticed and the correction work has started, the alarm is marked as noticed. It is possible to add a comment when entering the notice command. The comment is stored with the alarm in the alarm log.

Alarm Input/Output

The system can monitor external alarms via MGU or ALU board. With the use of this hardware external alarms (like fire alarms) can be reported in the alarm log of MX-ONE Telephony System. Alarms in MX-ONE Telephony System can also be connected to alarm outputs (that for instance can ring an alarm bell).

Alarm Log

If the alarm log becomes full, the alarms with the lowest alarm classes are removed from the log. A configuration parameter determines which alarms are least important.

System Availability

Alarms are not lost when the system is restarted. However, they are lost when the alarm feature is reloaded. After a reload, the feature will request are send of all alarms from the alarm sender.

SNMP

The Telephony Server has an *alarm and event handling* function. An SNMP-interface exposes standard MIB II information and alarms generated in the Telephony Server can be sent as SNMP traps. When alarms are cleared in the system, alarm cleared traps are sent to the connected SNMP Managers.

In addition to alarms, events are stored in the Linux syslog. These events are usually not critical for the operation of the system, but they might provide useful information on the behavior of the software, or when troubleshooting.

Automatic Fault Administration

The automatic *fault administration function* is used for the following when faults occur:

- Decide and carry out recovery actions to minimize the consequences for the system.
- Inform the system administrator about the faults and the results from the measures taken.

10.2.1.2

CMG

The CMG Server hosts a number of different applications that send events to the Windows Event Log.

10.2.1.3

OneBox

The Messaging Server hosts a number of different applications that send events to the Windows Event Log.

The Windows SNMP Agent allows external SNMP Managers access to standard MIB II information.

10.3 Configuration Management

This chapter describes how the MX-ONE components are used to configure MX-ONE.

Data in MX-ONE can be divided into the following data types:

- Telephony system data
- User and extension data
- Application specific data.

Telephony system data is managed using MX-ONE Manager Telephony System and comprises data regarding, for example, server configurations, media gateways, number plans, and routes.

User and extension data is divided into common and application specific data. Common data is managed using MX-ONE Manager Provisioning and comprises data that is used by two or more components, for example user data (used by MX-ONE Manager Provisioning and CMG), extension data (used by all components), and department data (used by MX-ONE Manager Provisioning and CMG).

Application specific data is managed using the management tool of the specific application, for example, CMG's *Directory Manager*.

10.3.1 MX-ONE Telephony System

MX-ONE Manager Telephony System is used to manage telephony system data, for example:

- Server configurations
- Number plans
- Common service profiles
- Groups
- Routes.

Initial configuration of MX-ONE Telephony System, including, for example, setting IP addresses of interfaces, is made during the software installation.

For detailed information, see the user guide for *MX-ONE MANAGER TELEPHONY SYSTEM*.

10.3.2 MX-ONE Manager Provisioning

MX-ONE Manager Provisioning is used to manage common user and extension data in MX-ONE, for example:

- Adding, changing and removing users

When managing user data in Manager Provisioning, the corresponding user data is automatically updated in the CMG and MX-ONE Telephony System databases.

- Adding, changing and removing extensions

When managing extension data in Manager Provisioning, the corresponding user data is automatically updated in the CMG and MX-ONE Telephony System databases.

- Adding, changing and removing administrators, security profiles, service groups, departments, locations, and subsystems
- Managing mail box services

When managing mail box services in Manager Provisioning, the corresponding data is automatically updated in the OneBox database.

- Importing user and extension data

Data can be imported from CMG, DNA, or using CSV files. The import function is used when, for example, performing a new installation of Manager Provisioning in an existing MX-ONE installation.

- Exporting user and extension data

Data can be exported from Manager Provisioning to XML or CSV files.

10.3.3

CMG Server

CMG Server is used to manage CMG specific data, for example:

- CMG specific user data, such as time zone settings. (Common user data is managed using Manager Provisioning.)
- Field name settings (all languages)
- Search layout settings for NOW
- Search layout settings for Office Web
- Settings of Activity codes
- Settings for system Contact Profiles including forwarding number
- Telephony connection settings
- Message Delivery Systems settings
- Settings of other system parameters
- Personal log-in
- Language settings
- Settings of other system parameters.

10.3.4

OneBox

The following are some examples of data that is configurable for OneBox Fax Mail:

- Settings for fax support on the Telephony Server
- Global fax settings
- Inbound fax routing
- Notifications about document processing and server status

For detailed information, see the getting started guide for *Captaris RightFax* and administrators guide for *Captaris RightFax*.

The following are some examples of data that is configurable for OneBox Voice Mail:

- Unified messaging settings
- SMS message notification settings
- Settings for networking Messaging Voice Mail Servers
- Settings for connecting the voice mail system to the telephony system

For detailed information, see the administrator guide for *Administering MX-ONE Messaging (OneBox Voice Mail Version 5.0)*.

10.4

Accounting Management

Accounting Management comprises the services provided by MX-ONE that can be used for billing purposes.

MX-ONE Telephony System generates customized Call Information Logging (CIL) records. Each record contains data for one call.

The CIL records can be configured to have several output formats, such as comma separated, Structured Query Language (SQL), or Extensible Markup Language (XML). Each Telephony Server generates its own CIL records that can be sent to up to 10 target locations. It is also possible to configure a single location and format for all CIL records.

The information provided by the CILs can be matched with user-specific information to get a complete picture of how each user uses the system.

For detailed information, see the description for *CALL INFORMATION LOGGING, QUALITY OF SERVICE LOGGING*.

10.5 Quality of Service

VoIP quality of service information can be logged per call. Delay, jitter, used codecs and packet loss rate can be logged. The data can be presented in Manager Telephony System or through Call Information Logging output.

For more information, see the description for *QUALITY OF SERVICE*.

10.6 Performance Management

MX-ONE provides the following level of Performance Management:

- Basic Performance Management (always included)

10.6.1 Basic Performance Management

The MX-ONE applications include SNMP agents as well as extensive embedded Performance Management functionality. Some of these functions rely on standard operating system components such as the Windows SNMP Agent, while other functions are embedded in the MX-ONE applications.

For a short summary of the basic monitoring and performance management capabilities of the MX-ONE servers and applications, see section 10.6.1.1 MX-ONE Telephony System and 10.6.1.2 OneBox.

10.6.1.1 MX-ONE Telephony System

MX-ONE Telephony System provides the following monitoring and performance management functions:

- Congestion on routes

The rate of congestion on routes is a critical resource in the Telephony Server and is continuously monitored. When the congestion rate reaches a certain threshold, alarms are generated.

- Traffic Measurements

Traffic measurements provide statistical information about the telephony traffic in the Telephony Server. Examples of such measurements are:

- Outgoing calls on a certain route
- Total number of incoming calls

Traffic measurements need to be configured and started manually and data will be collected based on a number of different criteria (configurable).

10.6.1.2 OneBox

The following three different tools are available for configuration in OneBox:

- Support Messaging System Management (a suite of Windows-based tools)
- Telephony Server Diagnostics
- Line Status & Reports

Note: The Telephony Server Diagnostics tool does not specifically refer to the component Telephony Server in MX-ONE and the tool is not part of MX-ONE.

10.7 Security Management

For detailed information, see chapter 11 Security, the description for *SECURITY*, and *SECURITY GUIDELINES*.

10.7.1 User Management

In MX-ONE Telephony System it is possible to define the users that are allowed to use the Command Line Interface and the access privilege for each of them.

10.7.2 Audit and Security Trail

The different components of MX-ONE log all activities that are relevant from the security point of view, such as access attempts, O&M operations, and so on.

This information is necessary in case of a security incident to be able to determine what caused the event.

10.7.3 IPsec

The servers used in MX-ONE Telephony System communicate using a proprietary inter-server signaling protocol, used for management messages, control signals, and call control signals. This communication can be protected by using IPsec.

10.7.4 Protection of VoIP

MX-ONE supports the use of secure real-time transport protocol (SRTP) in IP phones and MX-ONE Lite, MX-ONE Classic and MX-ONE Classic Stackable. EMG does not support SRTP.

MX-ONE supports Transport Layer Security (TLS), which provides secure access to IP phones and web services and secure signaling between IP phones and MX-ONE Telephony Servers.

11

Security

Attention to the security aspects of an IP telephony infrastructure is increasingly growing by corporate Chief Information Officers (CIOs), IT administrators, and users. Voice over IP traffic (both signaling and media) must be protected from a number of attacks, for example, media streams eavesdropping, toll-fraud attacks, and signaling modification. For this reason, it is necessary to protect both the VoIP signaling messages as well as the media streams.

The following security measures are supported in MX-ONE Telephony System:

- Secure RTP (SRTP) to protect media streams
MX-ONE supports the use of SRTP for media encryption in the IP phones and MGU and IPLU based gateways.
- Transport Layer Security (TLS) to protect signaling messages
TLS guarantees the signaling privacy when the SRTP keys are interchanged between the parties.
- Support for a number of flexible security policies, in order to support environment with different security requirements

The main principle for the security policy is that it directs if an extension is allowed to register to the system or not. Once the extension is registered, the calls to any other party is allowed from a security perspective.

SIP terminals have to authenticate themselves using HTTP digest authentication. If a PIN code is assigned to the user the authentication will also be done together with SIP request as the INVITE. To protect the communication between MX-ONE servers can IPsec be used.

The servers in MX-ONE run on operating systems that have been hardened to resist the most common network attacks. Known vulnerable services are shut down and file integrity is checked periodically. Additionally, customers are recommended to implement security policies that cover patch management and antivirus software updates. It is recommended to use some type of antivirus software and to have automatic updates, of the security patches, activated. To overcome the VLAN separation, server farms should be protected by firewalls and Intrusion Detection Systems (IDS) that are able to block attacks.

All management interfaces towards MX-ONE servers can be run over secure protocols, such as SSH and HTTPS. Management operations and access to such interfaces are logged to have maximum control. Users and administrators always have to authenticate themselves before being able to access the system. Additionally, an access control

mechanism is available to assign users and administrators different roles and privileges.

For detailed information about security, see the description for *SECURITY*.

12 Capacity

12.1 MX-ONE Telephony System

For information about the capacity of MX-ONE Telephony System configurations, see the description for *CAPACITIES*.

12.2 OneBox

Table 2 OneBox Voice Mail

Ports (simultaneous connected users)	Up to 384
VM users	Up to 40,000 users.
UM users	Up to 20,000 users

Table 3 OneBox Fax Mail

Ports	Up to 30
Users	No general guidelines as to how many users a 30-port system can handle, as fax usage varies greatly between customers.

For more information about the capacity of OneBox, see the relevant OneBox documentation.

13 Environmental Conditions

13.1 General

MX-ONE consists of different components as well as commercial server-based products.

MX-ONE is designed to operate in and comply with regulations in force for enterprise (light industry) locations. Special measures may be required if MX-ONE is installed in other locations or if the environmental parameters deviate from the values described in this document or in other referenced documents.

13.2 Climatic Environment

Table 4 Temperature Range and Humidity

Product		In Operation	Storage
MX-ONE Lite MX-ONE Classic MX-ONE Server Fan unit Media gateways supported by MX-ONE 5.0 but not available for new deliveries: Enterprise Media Gateway (EMG) Media Gateway Classic	Temperature range	+5° C to + 40° C IEC 60068-2-1, -2, -14 and -56; ETSI EN 300 019-2-3 Class 3.1, Table 1, Normal climatic limits	-5° C to +55° C IEC 60068-2-1, -2 and -56; ETSI EN 300019-2-1 Class 3.1, Table 1
	Relative humidity	+5% to 85% IEC 60068-2-1, -2, -14 and -56; ETSI EN 300 019-2-3 Class 3.1, Normal climatic limits (no condensation)	Maximum +95% ETSI EN 300019-2-1 Class 3.1 (condensation)
AC/DC power unit		As specified by supplier	As specified by supplier
Telephony Server and server for CMG and OneBox		As specified by supplier	As specified by supplier

Table 5 Mechanical Range

Product		Seismic exposure (product in operation)	Transport
MX-ONE Lite MX-ONE Classic MX-ONE Server Fan unit Media gateways supported by MX-ONE 5.0 but not available for new deliveries: Enterprise Media Gateway (EMG) Media Gateway Classic		VERTEQ II IEC 60068-2-6, -27, -64 and GR-63-CORE; ETSI EN 300-019-2-3 Class 3.2 Table 5	Random and bump IEC 60068-2-29, -32 and -64; ETSI 300 019-2-2 Class 2.2 Table 4
AC/DC power unit		As specified by supplier	As specified by supplier
Telephony Server MX-ONE CMG and OneBox		As specified by supplier	As specified by supplier

13.3 Electromagnetic Compatibility, Safety and Telecom

For Aastra products, see the Supplier's Declaration of Conformity, located at www.aastra.com.

For the Telephony Server, see the documentation delivered with the product or contact the supplier for details.

For safety information, see the description for *SAFETY*.

For the AC/DC unit, see the documentation delivered with the product or contact the supplier for details.

14 References

The following SIP RFCs are supported by the MX-ONE Telephony System.

RFC	Description
2246	Transport Layer Security (TLS)
2617	HTTP Authentication. Basic and Digest Access Authentication
2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
2976	The SIP INFO method
3261	SIP, Session Initiation Protocol
3262	Reliability of Provisional Responses in SIP (PRACK method)
3263	Locating SIP Servers
3264	Offer/answer method (SDP usage)
3265	SIP specific event notification
3310	Digest Authentication
3311	SIP UPDATE method
3323	Privacy mechanism for SIP
3325	Private Extensions to SIP for Asserted Identity within Trusted Networks
3326	The Reason header field for SIP
3398	ISDN User Part (ISUP) to SIP mapping
3407	SDP Simple Capability Description
3428	Extension for Instant Message method
3515	SIP Refer method
3550	Real Time Transport Protocol
3580	SIP event package for Registration
3581	Symmetric response routing
3665	SIP Basic Call Flow Examples
3680	SIP Event package for Registration
3711	The Secure Real-time Transport Protocol (SRTP)
3725	Third party call control in SIP
3842	Message waiting indication event package for SIP
3891	The SIP 'Replaces' header

RFC	Description
3892	SIP Referred-by mechanism
3960	Describes how to manage early media in SIP using two models: the gateway model and the application server model. Telephony Server supports the gateway model.
3966	The Tel URI for telephone numbers
3986	Uniform Resource Identifiers (URIs)
4028	Session timers in SIP
4040	RTP Payload Format for a 64 kbit/s Transparent Call
4235	Dialog event package
4244	An extension to SIP for Request History Information
4320	Actions addressing identified issues with the SIP non-Invite transaction
4346	Transport Layered Security (TLS) 1.1 and SRTP
4497	Interworking between SIP and ISDN QSIG
4566 (and 2327)	Session Description Protocol
4568	SDP Security Descriptions for media streams (SDS)
4733	RTP payload format for DTMF digits, Telephony Tones and Telephony Signals (note that RFC 2833 DTMF transport method 1, is not supported).
4904	Representing Trunk groups in tel/sip URIs
4975	The Message Session Relay Protocol (MSRP)
5626	Managing Client-initiated connections in SIP
5806	Diversion information in SIP
6337	Usage of the Offer/Answer Model

The following IP standards are also supported by the MX-ONE Telephony System:

ITU-T Recommendation	Relevant G-series recommendations (G.107, G.108, G.109, G.111, G.114, G.131, G.168, G.175)
ITU-T Recommendation	Relevant P-series recommendations (P.800, P.862)
ITU-T Recommendation Y.1541	Network performance objectives for IP-based services
ETSI ES 202 020	Speech Processing, Transmission and Quality Aspects (STQ); Harmonized Pan-European/North-American approach to loss and level planning for voice gateways to IP based networks
TSB 122-A	TIA/EIA Telecommunications Systems Bulletin Voice Gateway Loss and Level Plan Guideline