MiVoice MX-ONE

System Overview - Description

Release 7.1
April 9, 2019

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Introduction

MiVoice MX-ONE is a complete communications solution that is highly flexible to address the business communication needs in different vertical segments and sizes, scaling from 50 to 500K users in a single system. In this document the system will generally be called MX-ONE.

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. The combined solution addresses Enterprise communications need for all types of businesses in over 70 countries worldwide.

Standards Based Approach

MX-ONE is designed for use in an open standards based software and hardware environment, using industry standard servers with a LINUX SUSE & Windows server operating systems. Furthermore, together with strong support for open standards, such as SIP, CSTAIII/TR87 and Web services, it comes with support for a full range of network and user side interfaces, enabling connection to a variety of public networks as well as 3rd party systems and applications.

It also provides a strong migration capability, where existing MD110 and Mitel TSW customers can maintain a portion of already made investments and migrate to new technology and services at their own pace.

Fixed Mobile Convergence

Today's users are expecting to have all types of communication services combined and integrated in the same application GUI on the device of their choosing, whether in the office or when they are on the move.

MX-ONE sets the standard for Fixed Mobile Convergence 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.

The Fixed Mobile Conversion (FMC) is a smart phone application that is integrated with MX-ONE to provide Fixed Mobile Convergence. An FMC user is seen by the system as a normal SIP user thanks to the FMC controller, a software based Fixed Mobile Convergence gateway that inter-connects the mobile network (via SIP trunk) to the MX-ONE on the customer enterprise LAN. The FMC controller can reside as virtual machine or be delivered as an appliance.

An added feature with the FMC controller is the possibility for users to roam between GSM/3G/4G networks and the company Wi-Fi network with seamless handover in either direction (mobile phone dependent) during a call.

The FMC mobile apps for iOS and Android provide a rich graphical end-user interface for a variety of telephony services and offers features such as corporate directory look-up, activity call routing management and presence information.

The FMC also provides call routing without mobile operator support and advanced cost saving features (Dynamic LCR and traveling SIM) for international roaming users. For access to corporate contacts and rich presence information the FMC integrates with the Mitel BluStar Server (presence engine).

Unified Communications and Collaboration

With MiVoice MX-ONE 7.x the customer can benefit from native Unified Communication and Collaboration (UCC) through integration of Mitel's MiCollab portfolio, offering multi-media collaboration (voice, video and IM) through a variety of soft-phones and CTI call control clients for PCs, tablets and smartphones. These can be combined with the Mitel BluStar Ecosystem clients, as well as desktop SIP endpoints, ranging from the Mitel Conference & Video phone, to a full range of SIP terminals.

The MiCollab suite works as a SIP based client-server system which through standards based integration provides call control and feature-support natively by MiVoice MX-ONE.

Mitel's MiCollab portfolio provides businesses with a comprehensive, end-to-end UCC solution with a versatile and consistent user interface across different devices. MiCollab offers a natural collaboration experience and fully embraces trends such as presence information, IM, multi party and personal video communication as well as increased use of smart phones and tablets in the workplace.

MiCollab 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. Mitel's focus across the portfolio is a unified user experience, facilitating user-adoption through a consistent look and feel, regardless of the device used.

Mitel Application Server (MAS) is the core of the MiCollab suite. The MAS handles the clients, contacts, presence as well as control of, and signaling between, the MiCollab components and the MX-ONE system. All user-provisioning is handled through the MX-ONE Provisioning Manager and automatically executed via the MAS server.

Integrated with the MAS server the below MiCollab server applications are delivering UCC services to the end-users:

The MiCollab AWV (audio, web and video) suite multi-media meet-me web-based desktop sharing and video conferencing. Its web based conference administration portal allows users to setup and manage their conferences directly from any MiCollab client, or from any PC via web browser.

MiTeam, also part of the MiCollab portfolio, is a set of work-stream communications and collaboration tools

that provide a highly collaborative, persistent workspace for team-based meetings, conversations, content collaboration and project management.

Mitel NuPoint Unified Messaging (UM), which provides messaging functions from basic voice mail to advanced unified messaging.

MiVoice Border Gateway (MBG), a platform for secure communication with remote tele-workers, can also be used.

Collaboration Management - Managing How You Want To Be Reached

Another aspect to consider is Collaboration Management or the possibility to manage how calls are routed based on a user's presence or activity.

The Mitel Collaboration Management Suite (Mitel CMG) offers solutions to fully automate this process by integrating into a company's calendar system and setting diversions or personal number routing profiles for a user, based on their calendar activities.

The system comes with a Web based Self service portal (Mitel BluStar CMG Web) that allows users to set these call routing options up based on their preferences. Once they are setup, the user no longer has to worry whether or not his diversions are set or if customer calls are being handled in his absence. He can even change them on the fly, through a simple press of a button, if something urgent comes up that was not planned.

The web portal also offers other services, such as directory search, presence and line state and click to call. It is designed using responsive web technology, enabling it to be used on any device - PC, Mobile or tablet - regardless of location. This means that even out of the office, CMG BluStar users have access to their services on any mobile device with a browser. Additionally, it integrates with other parts of the portfolio, such as BluStar for PC or FMC as an embedded link.

The CMG Virtual Reception option provides Auto-attendant, VM and reception/visitor services. These are value add services that complement the Collaboration Management suite to provide users with a complete suite of services that provide enhanced customer services. If Mitel Virtual reception is present with the CMG deployment, Mitel BluStar CMG Web users will also have access to extra capabilities in the self- service web portal, including management their personal IVR settings, visual VM and visitor management services.

Attendant Services - A Company's Face To The World

When professional attendant services are needed with MX-ONE, Mitel InAttend is the platform of choice. It is a Windows based client-server architecture than integrates with a customer's Microsoft Active Directory and calendar systems. Attendants have a full featured PC based client that offer all the expected features for a professional attendant, including SIP soft-phone (or hard-phone) with advanced attendant call handling features (keyboard or mouse), multiple queues, A/B panels, advanced directory search, view users' presence, line state and calendar activities, messaging, etc. In other words, everything an attendant needs to provide fast and efficient service to customers.

The Mitel BluStar Server (Presence engine) is embedded in the solution enabling the attendants to get real time presence and line state, It uses industry standards based interfaces such a SIP/simple, CSTAIII and UCMA to provide rich presence information from different systems, including MX-ONE, Cisco CUCM and Microsoft Lync 2013/Skype-for-Business. This makes sure the attendants have federation of all rele-

vant status information in one place, regardless of the system users are connected to. The key benefits are efficiency and better customer service.

Furthermore, Mitel InAttend has tight integration with Mitel CMG, enabling it to see and even manage user presence/activities and diversions. Mitel InAttend is a scalable solution that can start with one attendant for medium sized companies, yet scale to large organizations with 100 or more attendant consoles.

One Solution Addressing Different Business Models

The MX-ONE design and license models are built to work either as Customer Premises (CPE) deployment or as a Cloud or hybrid Cloud deployment. Its license model is built up to enable support for a either a traditional Capital Expenditure (CAPEX) model or OPEX based subscription models. The license reporting software embedded in the system allows accurate and secure license usage monitoring needed for a subscription based licensing model. The system license usage information is reported on a monthly basis, enabling a "pay for what you use" model.

The MX-ONE Private Cloud offering provides customers the same user-experience as a traditional premises-based deployment and only differs in the license delivery model.

The following deployment variants exist for the MiVoice MX-ONE solutions:
- MiVoice MX-ONE CPE Turnkey solution deployment. For large (and medium) CPE solutions.
- MiVoice MX-ONE CPE Software only deployment. For large (and medium) CPE solutions.
- MiVoice MX-ONE CPE Virtual appliance deployment. For large (and medium) CPE solutions.
- MiVoice MX-ONE Express, single server pre-packaged smaller solution (30-1500 users) with UCC and FMC, running on an ASU-II server, or as "Software only" for a VMware deployment. A wizard simplifies installation and configuration.

   NOTE: The MiVoice MX-ONE Express is not supported from MX-ONE 7.0 release onwards.
- MiVoice MX-ONE MLA (Managed Services License Agreement), cloud deployments with feature based user licensing. Includes UC and FMC, and a wizard to simplify installation and configuration.

Management System

The Management solution continues to grow 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 Service Node call server and UCC applications offering HW consolidation and enhanced resiliency options.

NOTE: Note:Re-branding to Mitel names was done in the MiVoice MX-ONE 6.x, but some Aastra related names are still used for specific legacy products, so a few such names may exist later in this document.

System Architecture

General System Overview

An overview of the components that comprise the MiVoice MX-ONE System is given in .

Figure 1: System architecture brief overview
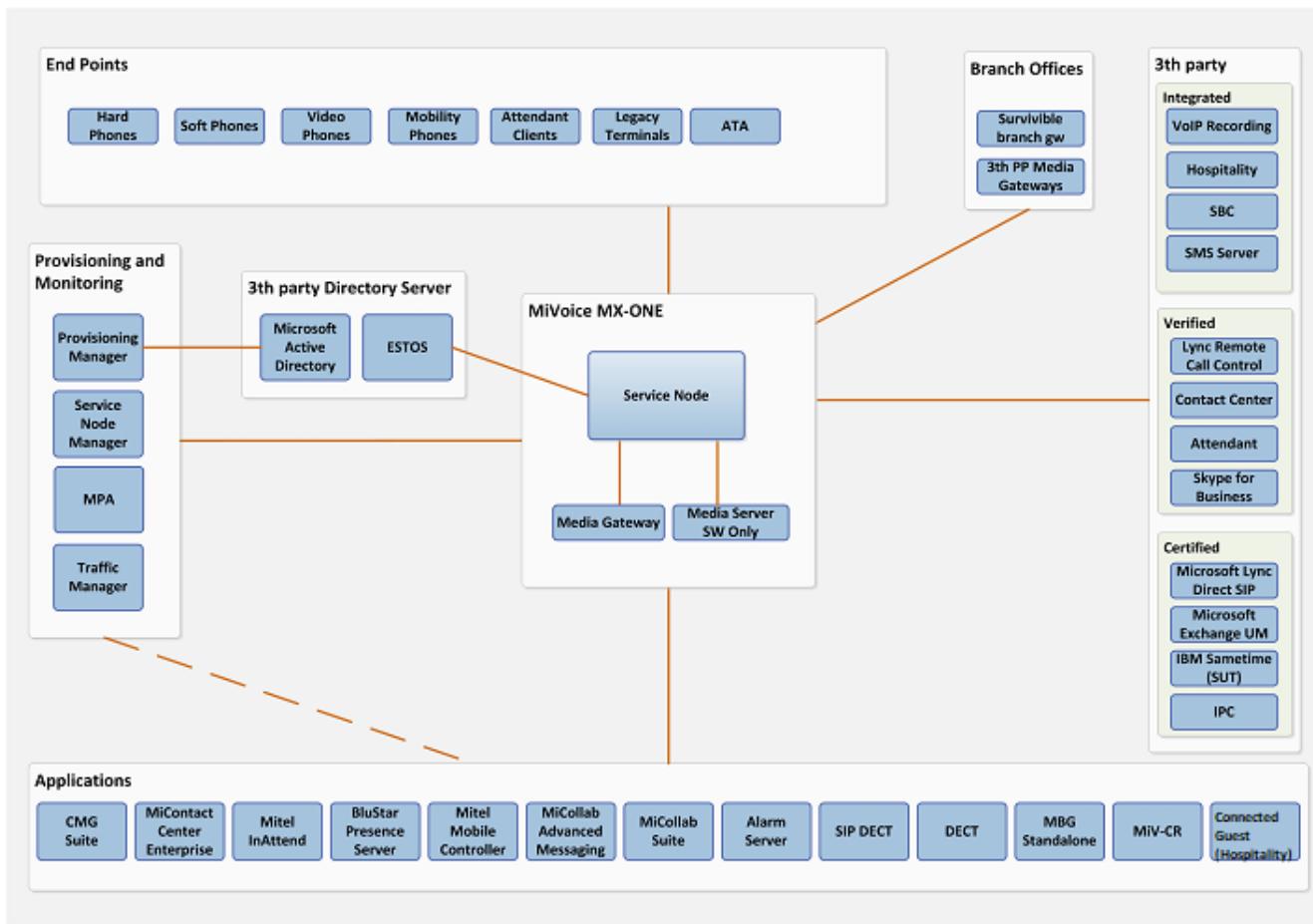
MX-ONE Communication System Components

The MiVoice MX-ONE communication system is comprised of the following three main components:

- The call server component that takes care of the signaling is called **MX-ONE Service Node** (former Telephony Server), which is a Linux based call control software that can either be installed in a private cloud as a instance or reside in a standard Intel based server.

- The second component is a software based Media Server with DSP resources for handling tone detection, multi-party conferencing and packet switching between different IP end-points (SIP only environment), when direct media is not possible. In SIP only installations, there is no need for dedi-

cated media gateway hardware. The Media Server has a load-sharing/load-balancing functionality for such installations. The media server can reside in the same Linux machine as the Service Node call server, thus reducing the customer footprint, but can also reside on a separate server.

- As a complement, one or several hardware based Media Gateways can be added to the configuration, to provide the physical interfaces towards TDM subscribers, public networks and Auxiliary devices. It also houses DSP resources for handling tones, conferencing, packet switching towards IP Phones (SIP and H.323) and to convert media between different protocols. Any combination media servers and media gateways up to a limit of 15 can be connected to the same server.

Other MX-ONE System Components

Mitel MiCollab portfolio, is designed to provide business with effective multi-media collaboration amongst its employees, partners and customers. The portfolio includes:

- Mitel MiCollab Application Server, the core component of the Mitel MiCollab system, providing directory, presence and calendar integration for end users. The MiCollab application Server also provides a homogeneous environment for deployment and operation of the MiCollab environment and integrated with MX-ONE for smooth and consistent user population and configuration.

- Mitel MiCollab clients comprises a Unified Communications SIP soft phone/call control client application for PC as well as mobile apps for IPad/iPhone, Windows mobile and Android that enable corporate users to make and receive voice and video calls over MX-ONE via SIP and to interact with other users and MiContactCenter Enterprise agents through IM (Instant Messaging), e-mail, voice and video.

- MiCollab Audio, Web, Video (AWV) is a powerful meeting point and conference server for various forms of collaboration.

- MiTeam, a set of work-stream communications and collaboration tools that provide a highly collaborative, persistent workspace for team-based meetings, conversations, content collaboration and project management.

- Mitel NuPoint Unified Messaging (UM)

- MiVoice Border Gateway (MBG)

MiVoice Conference and Video Phone, is a compact audio and video conference room unit offering an HD quality user experience. It integrates fully with the MiCollab AWV conferencing suite for voice, video and content sharing. Additionally, it can host a 4-way video conference and inter-operates with Mitel MiCollab end-points and Mitel BluStar Client end-points.

MiCollab Advanced Messaging (former OneBox) provides a powerful suite of Unified Communications applications including advanced call processing, Auto-Attendant, voice mail, e-mail integration, personal assistant, fax, speech driven services, notifications, etc.

MiContact Center Enterprise (former Solidus eCare), a suite of applications and services for server-based contact centers.

MiCC Enterprise offers an enterprise-wide contact center solution with IP and mobility across multiple sites. This enables distributed customer service organizations to behave as one single unit.

Mitel Collaboration Management Suite (CMG)

- Mitel BluStar CMG Web is a set of services that enables business users to manage their day-to-day activities. With the Mitel BluStar CMG Web, users can do "smart-search" directory services, click-to-dial, manage activity timeline and their call-routing preferences based on their calendar/activities. The integration with the BluStar Server presence engine enables users to see in real-time their colleagues' rich presence information, including BluStar clients presence status, calendar activity and line state from all available sources provided by the BluStar Server.

- Mitel Virtual Reception - Automated speech and reception services.

   Mitel Virtual Reception is a set of automated self-services that significantly reduce wait time and attendant workload, increasing at the same time customers' flexibility and efficiency.

Mitel BluStar Ecosystem, providing multi-media services for end users.

- Mitel BluStar Server, the core component of the BluStar Ecosystem, providing presence and calendar integration for end users.

- Mitel 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 Mitel's MiContact Center agents through IM (Instant Messaging), e-mail, voice and video.

- BluStar 8000i and BluStar for Conference Rooms.

Mitel Mobile Client (FMC, former AMC), a smart phone application that makes the phone an integral part of MX-ONE.

MiVoice Call Recording is an application for recording of VoIP calls.

Open Integration Gateway (OIG) is supported, and offers a dedicated back-end API to enable CTI call control and Click-to-dial for various applications, for example MiCollab and MS Skype-for-Business clients.

MX-ONE end-points supporting all various types of extension interfaces (SIP, H.323, analog, digital, DECT, Wi-Fi and others).

Mitel Border Gateway (MBG), for secure remote access from MiCollab clients. Additionally the MBG can provide security for SIP trunks to public networks or remote application servers, and facilitate VoIP recording from the gateway in cases where the other end-point does not support recording.

MX-ONE Management applications, with MX-ONE Service Node Manager, MX-ONE Provisioning Manager, and Mitel Performance Analytics (MPA). See *Single-point-of-entry Management*.

For information about the different components/applications, see their respective descriptions.

System Architecture

The long proven scalable and flexible architecture of MX-ONE, allows a customer to tailor make an optional solution for the customer's specific needs. An MX-ONE solution can be built as a single central system with multiple servers, a distributed system database, and multiple gateways, but it also allows distribution of the system components over a Wide Area Network. Multiple MX-ONE systems can be networked with intelligent signaling systems in order to support several hundred thousand users within a global voice network.

Scalability - MX-ONE allows organizations to easily expand its capacity, add functionalities, modules, gateways, and easily upgrade the servers. A single MX-ONE can scale from 50 users 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 and Media gateways within a system can be spread over a wide area network in order to create solutions for small and large customers across geographical locations or as a centralized system with the servers on a central location.

Cloud Architecture

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

Cloud Computing broadens the horizons across organizational boundaries of "re-usability 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 deploy virtualization.

With MX-ONE it is now possible to run the MX-ONE Service Node 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 native Resiliency and High Availability options.

Furthermore, MX-ONE can optionally be installed as a Soft-switch without any hardware based media gateways for pure IP environments. In a virtualized environment, both the call server and media server reside in the same virtual machine and can benefit for VMware's High Availability options.

High Availability

Redundancy Functions

MX-ONE offers high availability by supporting different types of redundancy to cater to different types and requirements of the Enterprises.

MX-ONE offers far-reaching alternatives for High Availability.

Server redundancy delivering high availability:
- Home Location Register (HLR) redundancy. In this case, you are deploying IP/SIP phones and have more than one server sharing the load. The user data is synchronized between the servers. Should one of the servers fail, the IP/SIP phones will re-register to one of the secondary servers (using a temporary guest/backup-HLR) and recover most of their features. When the primary server comes back, they will re-register to their primary HLR server.

- 1+1 server redundancy and network redundancy. This is a scenario where a primary server and a dedicated back-up server are configured in a redundancy cluster. Should the primary server fail, the back-up server will take over. This is a warm stand-by setup.

- N+1 server redundancy and network redundancy. This scenario is similar to the 1+1 scenario, except you have several servers in the same location in a redundancy cluster with a dedicated back-up server that will take over if one of the primary servers fails. This is a warm stand-by setup.

- High Availability provided via virtualization software (e.g. VMware HA/FT). The MX-ONE Service Node fully supports VMware HA scenarios - both High Availability (Warm Standby) and Fault tolerance (Hot Standby) options. It assumes the proper virtualization infrastructure is put in place according to VMware specifications.

Network Redundancy with Bonding/Link failover is supported for the media gateways. Dual sub-net is no longer supported.

Branch Office Survivability

For Branch office situations where there are requirements for local PSTN access and survivability for IP/SIP terminals, sites can be equipped with a remote server and media gateway combination - known as a Survivable Branch Node (SBN).

The SBN bundles include the necessary trunk, tie-line and user licenses as well as necessary HW, depending on the options or package selected. As it is using standard MX-ONE Service Node software, it can be configured and managed centrally from the MX-ONE Management suite as a remote networked node.

True Mobility

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

Mitel 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 one 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 on 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.

SIP - The basis for Unified Communications and Collaboration

Session Initiation Protocol, SIP, has become the signaling protocol for establishing real-time multimedia communications. SIP is key to accelerating the transition to Unified Communications in a modern organization.

MX-ONE provides a high level of functionality with both SIP extension and SIP trunk with support for more than 45 RFCs. Mitel will continue to develop this architecture in order to provide our customers a natural migration path towards Unified Communications and Collaboration. As an example, MX-ONE SIP extensions now offers a higher level of functionality than what was previously offered with proprietary digital phones.

These same standards are also the basis for integration with 3rd party UC applications and an organizations business processes as well as the connection to the outside world with SIP trunks as an alternative to traditional PSTN networks.

Native SIP implementations in MX-ONE

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. Mitel 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-ONE 7.x 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 four active (simultaneously registered) multi-media SIP end-points.

SIP trunking - MX-ONE is certified for a number of SIP network providers and conforms to the SIP Connect 1.1 standard. MX-ONE's SIP trunking interface is also the basis for integration with 3rd party systems, such as Microsoft Lync 2013/Skype-for-Business, IBM Sametime SUT and other platforms to enable federation between different communications platforms.

Furthermore, with MX-ONE 7.x, a SIP tie-line with full network services is supported (introduced in MX-ONE 6.0), enabling feature transparency between MX-ONE nodes and certain applications, such as Mitel InAttend.

Single-point-of-entry Management

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.

Figure 2: System Management architecture overview
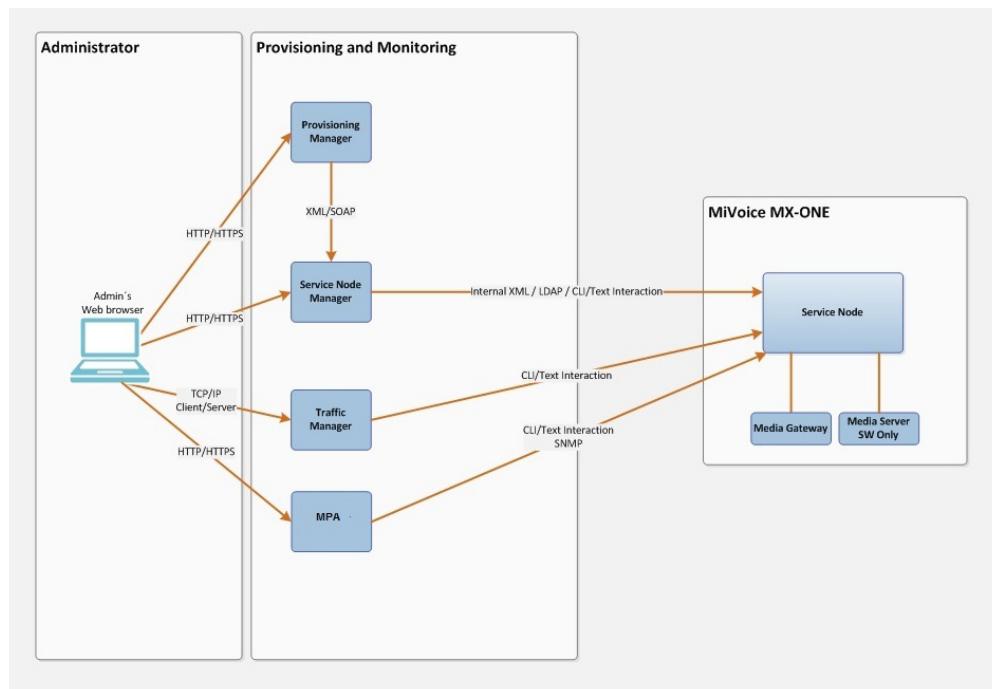
MX-ONE offers the following web-based managers to fulfill the administrative, provisioning and performance monitoring on the same place for MX-ONE, UC applications and terminals.

• MiVoice MX-ONE Service Node Manager

• MiVoice MX-ONE Provisioning Manager

• Mitel Performance Analytics (MPA) (former MarWatch)

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.

Mitel Performance Analytics (MPA), provides fault and performance management for multiple enterprise VoIP systems and associated network infrastructure, both LAN and WAN. MPA supports monitoring and remote access both for private networks, such as enterprise LANs and VPNs, and for public network or Internet-reachable devices, such as access routers.

Telephony Features

This section describes a few prominent features supported by MX-ONE.

For more information about the many telephony features and their capacities, see *MIVOICE MX-ONE FEATURE LIST* and *CAPACITIES*, but also specific descriptions for the features.

Extension Features

Multiple terminal services (Forking, Parallel ringing)

The Forking service, available for generic extension types (IP, DECT and mobile extensions), provides the user with one single number simultaneous ring signal on two to four predefined answering positions for an incoming call to the user. When the user answers the call, the call is directed to the terminal where it has been answered.

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 terminals might not ring at the same time depending on their type.

Parallel ringing supports analog, digital, CAS, cordless, IP, IP DECT, WiFi 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.

Note that the terminals in a parallel ring list must have different extension numbers, a main number which belongs to the user, and secondary numbers for the additional devices. The user's main number is the number that is used (in the directory) and seen by others, irrespective of which terminal that is used for a call.

For more information, see the operational directions for *MULTI-TERMINAL SERVICE*.

One Directory Number - Multiple SIP Terminals

An extension user can log on to the same extension number on several SIP terminals, i.e. do "forking" 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 "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 MiCollab or BluStar soft clients and the BluStar 8000i and benefit from the same video license. With the true

mobility provided by MX-ONE, a video authorized user can log-on to any video capable terminal, e.g. a BluStar 8000i or BluStar for Conference Room.

Supported end user terminals, any choice up to a total of four:
- Mitel 6900/6800/6700 desktop SIP terminal
- MiCollab client for PC, and for mobile phones
- BluStar for PC
- BluStar 8000i Desktop Media Phone
- Remote extension over SIP, i.e. mobile (with or without FMCC)
- 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 one SIP user license and one mobile extension license, and up to three additional device licenses, a total amount of five licenses.

Take call on other terminal

From MX-ONE 6.0 there is a new feature for users of multiple SIP terminals registered on the same number; the "Take call on other phone", which is a procedure to pick an ongoing call over to another of the own terminals. The ordinary group call pickup procedure (e.g. *8#) is used.

Extension Monitoring, Shared Call Appearance

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.

For SIP terminals the Shared Call Appearance (SCA) feature is supported as a monitoring service alternative to the MNS.

For more information, see the description for *SHARED CALL APPEARANCE* and the operational directions for *BOSS-SECRETARY*.

Figure 1:   Example of a Manager-Secretary Solution
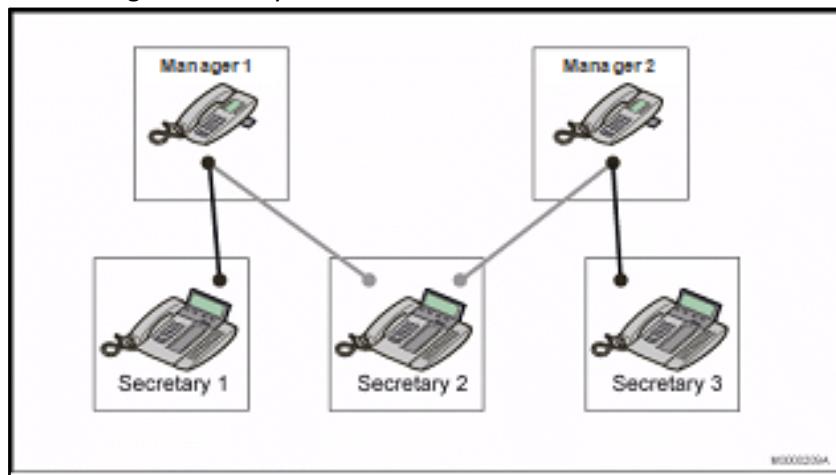
Personal Number

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 calender and location information stored for the user.

A CTI group number can be stored as a number in the personal number list.

For more information, see the description for *PERSONAL NUMBER/CALL LIST*.

Boss-secretary Configurations 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.

For more information, see the description for *PERSONAL NUMBER/CALL LIST* and the operational directions for *BOSS-SECRETARY*.

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 Mitel 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 Application Connectivity Server (ACS). The ACS is connected with SIP to the MX-ONE. The ACS provides queuing and call distribution. The use of ACS and SIP connection makes it possible to offer Mitel InAttend also to customers with PBXs from different vendors, who still want a uniform attendant functionality.

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 applied to MX-ONE 5.0. Furthermore, with MX-ONE 5.0 some net services were made available for this SIP trunk connection between MX-ONE and ACS to enable certain feature interactions between MX-ONE and the Mitel InAttend attendant suite. In MX-ONE 7.x additional services are supported, and service profiles are used.

Configuration changes to the SIP trunk setting must be done in both MX-ONE 7.x and ACS to enable enhanced feature interaction.

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

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*.

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.

SIP terminals that are group call pick-up members, can also support display of caller information for waiting group call pickup calls, and get a simpler pick-up procedure.

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 selection, with evenly distributed selection, or with parallel ringing (cascade ringing) on a limited number of members.

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, or by using a dedicated log-off procedure.

There are service procedures for a member's login/logout of the group (also valid for cascade ring groups). For members with SIP phones, there is also group member status indications.

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

Cascade Ring Group

From MX-ONE 6.0 there is support for a new variant of hunt group, with ringing in parallel on up to 16 extensions per group. All extension types supported in ordinary hunt groups are also supported in cascade groups. ADN and EDN numbers are not allowed as group members. Nor is MNS or MDN features allowed on a cascade ring group member.

There are service procedures for a member's login/logout of the group (also valid for ordinary hunt groups). For members with SIP phones, there is also group member status indications.

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 diverted 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.

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.

Outbound call center traffic is also supported.

Private Network Features

MX-ONE can be included in a private network with other MX-ONE systems or Mitel TSW/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 as 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 (SIP 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 2:   Private Network Features
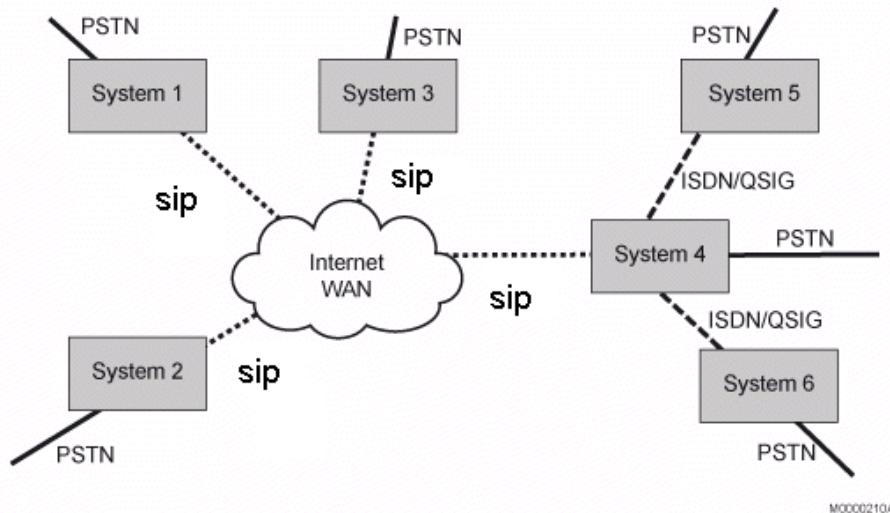
The following documents include more information:
*   Description for *SIP PRIVATE NETWORKING*
*   Description for *SDN, H.323 AND DPNSS NETWORKING*

Network Features

In MX-ONE, a number of common network features are supported.

Typical network features are Diversion, Callback, Intrusion, Call Waiting (call offer) and Bypass diversion. 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 request/affect user features, not system features.

For connections with SIP tie-line, system features as well as user features are applicable.

For connections with H.323 IP signaling, system features as well as user features are applicable.

For connections with ISDN, system features as well as user features are applicable.

The network features for ISDN are divided in those using standard QSIG signaling and others using proprietary signaling.

Other types of network signaling are also supported, for example DPNSS.

When MX-ONE is connected to a Mitel TSW/MD 110, all features are available for DPNSS, ISDN and H.323 signaling, but SIP is not supported.

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 3:   Centralized attendant
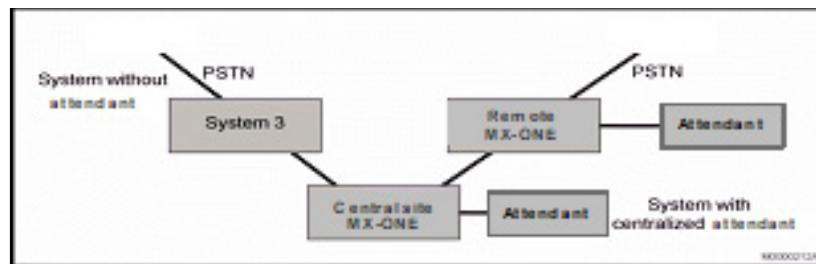
For more information, see the operational directions for *CENTRALIZED ANSWER POSITION*.

Least Cost Routing

The Least Cost Routing (LCR) feature enables MX-ONE 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 if that is the case, the call is routed to a PBX within the private network as close to the public destination as possible.

The following four situations can occur:

1) The dialed destination shall be routed directly to the public network.

2) The dialed destination is an extension within the own system (i.e. off-net to on-net routing).

3) The dialed destination is in the private network and can be reached completely through the private network (off-net to on-net routing).

4) The dialed destination can be reached partly through the private network (tail- or best-end hop-off).

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

Alternative Routing

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

If the primary route is not available, or if a class of service does not permit the call on this route, 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 4: Alternative Routing
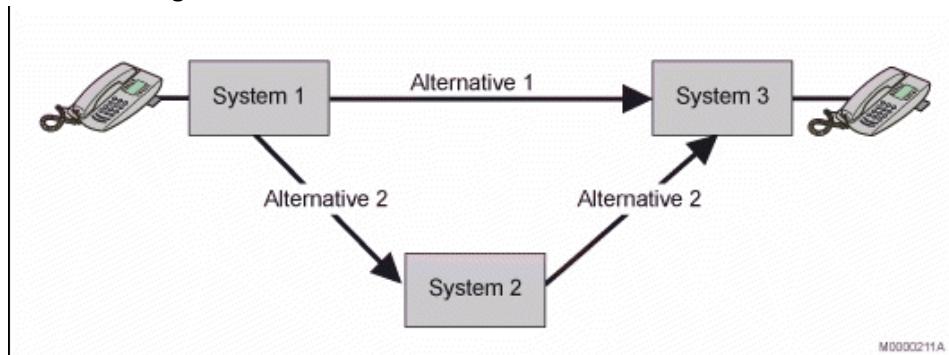
Multi Tenant functions, native and virtual solutions

From MX-ONE 6.0 Multi Tenanting is supported, both in a native variant, and in a virtualized variant using VMWare Virtualization, see *Virtualization*.

The functionality utilizes features like:

- Independent number plans per tenant
- License usage reports (for billing)
- Feature packaging with license control
- Capacity enhancements for certain features, like number of customers, hunt groups, recorded announcements, conferences, MNS keys, SCA keys.
- Enhanced and new extension group functions
- Remote provisioning of end user clients and phones
- Modified Call Discrimination (that can block inter-tenant traffic)
- New/modified Management functions supporting multi tenant and different authority levels

Customer Group

General

The customer group feature is used for two significantly different applications; for departments within one enterprise, or for different enterprises/customers, i.e. tenants, sharing one system. Note that the terminology has been changed in this revision of the system description.

Functionality

An MX-ONE can serve end users belonging to different departments within an enterprise, or basic hosting customers, that can be completely separated regarding certain services. For example Call Discrimination can force the extensions to use routes/trunks to reach a user in another department (customer).

Departments (customers) have their own resources, such as routes, recorded announcements, and attendant groups, but also have features for using certain common resources within the system.

Figure 5: Customer Groups
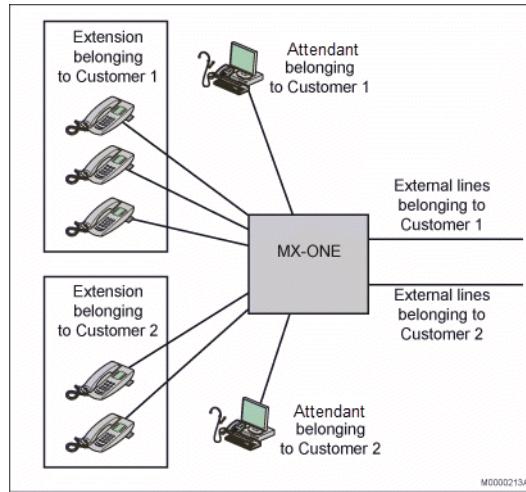
A department (customer) can be assigned to a group of attendants, a day or night service answering position, outgoing routes, and incoming routes.

The department (customer) group feature makes it possible for the users in a department (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.

The department (customer) group feature includes the following 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
*   Recorded announcements

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.

The complete MX-ONE solution is based on MiVoice MX-ONE, with the Hospitality license. combined with either Mitel ConnectedGuest or an external Hospitality Middleware certified to work with MX-ONE.

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.
- To obtain the specific Service Quarter functionality, the SQ phone must be Digital.

  If a SIP phone is used in the Service Quarter, the same functionality is obtained, except that the extra guest information is not shown.

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

VoIP Recording

With active VoIP recording, either third-party recording applications, or the MiVoice Call Recording application, the recording system will order the terminal to make 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 dependent 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.

Figure 6: Active IP recording
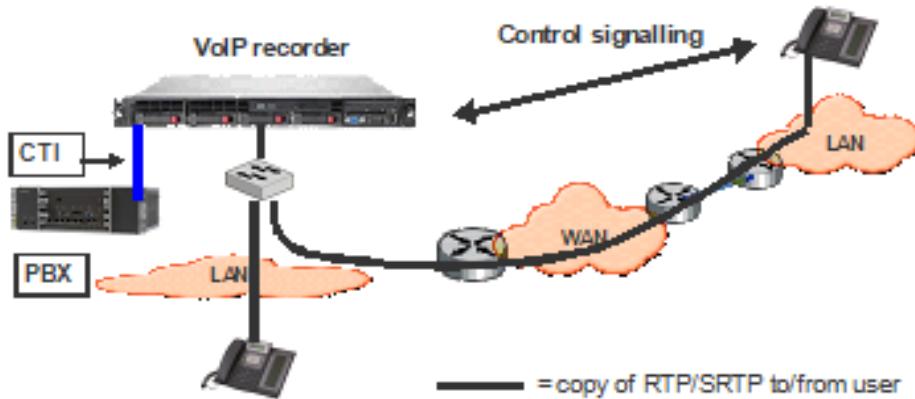
Active VoIP recording relies on integration with the PBX as the recording system must know the actual IP address of the users terminal when ordering a terminal to start recording. Furthermore, if encryption is used, the recording system must get the encryption keys. The integration is normally using SIP between the recorder and the terminal and CTI between PBX and recorder.

MX-ONE supports Active/Record-on-Demand recording of encrypted calls in combination with 6900/ 6800/6700 SIP phones and 7400/MiVoice 4400 H.323 phones. Encryption key handling is supported over TR87/CSTA3 and CSTA1 via Application Link.

Trunk side recording is also supported, for certain TDM E1/ISDN trunks (with additional HW) and for SIP trunks.

The Mitel Branch Gateway, MBG, can also be used, for VoIP recording on the gateway side, in case the other end-point (phone or soft-client) does not support recording.

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.

There are several supported implementations and configurations of the RVA feature, either with wav files stored in HW (MGU based) or with media streaming via Media Server (SW based).

Name and Progress Presentation

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

Currently progress information can be presented in most Latin alphabet languages as well as in Russian. The language to use can be configured for the system, but also for individual extensions and attendants.

For calls from public subscribers, where a calling party number is available, there are optional functions for access of public subscriber names. The MX-ONE will query external databases for a name, based on the number, and (if found) display the name on the called extension's display.

Blacklisting of Calling Public Subscribers

For calls from public subscribers, where a calling party number is available, there is an optional function for blacklisting, i.e. blocking the caller from making calls to any user in the PBX. The MX-ONE may provide a voice announcement (a vocal guidance), but will then disconnect the call, or possibly, if configured, redirect the call to a predefined destination.

Streaming on Idle Extension

For Mitel 6900/6800 SIP terminals there is a feature for streaming of voice media to the phone, when it is idle, i.e. not involved in a call or otherwise in use.

The feature requires a Media Server specifically configured as a streaming server supporting a SIP/MSCML protocol (according to RFC 5022). It also requires a dedicated key on the SIP phone, which provides a menu for the Streaming on idle service.

MiVoice MX-ONE Service Node Software

This chapter describes the MX-ONE Service Node software.

General

The same Service Node 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 *Service System*.
- Telephony applications
- MX-ONE Service Node Manager (*MX-ONE Service Node Manager*).
- MX-ONE Provisioning Manager (*MX-ONE Provisioning Manager*)
- MX-ONE Media Server

Operating System

The operating system is Novell SUSE® Linux Enterprise Server (SLES) version 12.

From MX-ONE 7.0, the 64-bit SLES 12 Linux Operating System version is used in the MX-ONE Service Node and in the Management applications. The SLES 12 provides some enhancements of redundancy, installation and upgrades.

Virtualization

The MX-ONE supports larger customer multi-server environments, which offer 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 focus for virtualization with the MX-ONE solution is to consolidate the number of servers and to limit the number of physical servers necessary for the complete MX-ONE solution, including the MX-ONE call manager and all 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). It also supports Hyper-V virtualization.

MX-ONE Service Node 7.x also supports this, and can run in a virtualized environment, formally named Virtualization. Currently MX-ONE relies on VMware's vSphere 5.x software suite as part of its virtualiza-

tion solution. For details, see *VIRTUALIZATION DESCRIPTION* and *MX-ONE SERVICE NODE VIRTUALIZA-TION DESCRIPTION*.

Native Support of IPv6

From MX-ONE 6.0, the system supports IPv6 addresses, in addition to IPv4 addresses.

The MX-ONE Service Node call manager and gateway components can operate in a native IPv6 network. It assumes the platform HW and OS software is able to operate using IPv4/IPv6 "dual stack" interfaces to the IP network. The alternatives are IPv4/IPv6 dual stack or IPv4 only.

The software components of the MX-ONE elements (call manager and gateways) can be configured to perform and interwork using standard IPv6 addressing from a signaling and media perspective.

The inter-server communication will if IPv4/IPv6 "dual stack" is installed, always use IPv6.

The entire system (all servers) must use the same IP version.

For SIP terminals, client and trunks, both IPv6 and IPv4 are supported. For H.323 terminals, client and trunks, only IPv4 is supported. If one terminal only supports IPv4, and another terminal only supports IPv6, a call between these two terminals will have to be a gateway connection.

The MGU and Media Server support both IPv4 and IPv6, but for the MGU, IPv6 only if security (signaling encryption) is not used. Older media gateways only support IPv4.

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 *MIVOICE MX-ONE FEATURE LIST*.

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, 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*.

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

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.

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.

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.

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

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, MX-ONE Lite and MX-ONE Slim 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.

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 - Service Node (for example adding a new server to an existing system)
    - MX-ONE - Media Server (soft media gateway)
    - MX-ONE - Media Gateway "1U"
    - MX-ONE - Media Gateway Lite ("3U")
    - MX-ONE Classic ("7U" with MGU)

    From MX-ONE 6.0, it is possible to have MX-ONE with a new version of the Media Server, which has higher capacity, and can be mixed with HW based Media Gateways. Codecs G.729 annex a&b are supported by the Media Server.

- Logical units
    - Program unit

Start and Restart of MiVoice MX-ONE

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.

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.

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.

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 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, 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 Service Node 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.

License Handling

The license handling in the MiVoice MX-ONE is primarily that licenses are required for all types of end-point resources, like extensions, attendants and trunks. For example, extensions and trunks have separate licenses, feature package licenses can be used (or must be used in MLA (Managed Service License Agreement) systems), and specific licenses exist for group function and other optional service related entities. There are individual licenses (per user or resource) and system licenses (per system).

A license usage report function, which can periodically report the number of used licenses, is also supported.

Server number 1 of a multi-Server 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. There are two types of license files, trial license (time limited) and real license files. From MX-ONE 7.0 there is a new version of encryption of the license file.

The following two license types exist:

• Individual license

  The individual license is used for ports, for example, the SIP extension or 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 (via O&M). 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.

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

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.

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.

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.

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

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, MX-ONE Lite and MX-ONE Slim 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 da-
ta, 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 cur-
    rent 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 Server (Service Node) has a disturbance counter that reflects the quality of the program execution
in the Server. 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.

Alarm Handling

Alarm handling is handled by the service system. For more information about the alarm handling func-
tion, see *Fault Management*.

Automatic Fault Administration

Automatic fault administration is handled by the service system. For more information about the auto-
matic fault administration function, see *Fault Management*.

Signal Tracing

The signal tracing function implements both the trace error log and the general trace function. The func-
tion 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. For tracing on MGU/MGU2, see also the MGU COMMAND AND FAULT LOCATION GUIDE on Mitel Knowledge Base.

Integrated Components/Applications

The following functions are separate products that are integrated with the MiVoice MX-ONE 7.x system. The listed applications are just examples, not a complete list.

Micollab Portfolio

The MiCollab portfolio contains the following entites:

- **Mitel MiCollab** client comprises a Unified Communications SIP soft phone/call control client application for PC as well as mobile apps for IPad/iPhone, Windows mobile and Android that enable corporate users to make and receive voice and video calls over MX-ONE via SIP and to interact with other users and MiCallCenter agents through IM (Instant Messaging), e-mail, voice and video.
    - Support for both Ad-hoc and scheduled conferences
    - Audio Conferencing – up to 300 users in one conference
    - Web client for conference host and remote/guest participants
    - Desktop sharing from PC
    - Multi-party video
    - Integration with MiVoice Conference phone for collaboration and video sharing
- **Mitel Application Server (MAS)** is the core of the MiCollab suite. The MAS handles the clients, contacts, presence as well as control of, and signaling between, the MiCollab components and the MX-ONE system. All user-provisioning is handled through the MX-ONE Provisioning Manager and automatically executed via the MAS server.
- **Mitel MiCollab (7.x) AWV (audio, web and video)** is a meeting point and conferencing component of Mitel's Unified Communications and Collaboration (UCC) portfolio, MiCollab, that is designed to provide businesses with effective collaboration amongst its employees, partners and customers. The MiCollab Conferencing AWV is an integral part of the MX-ONE solution offer. AWV provides a dynamic, real-time environment for the sharing of ideas and information. It allows employees throughout the business to connect with people, both inside and outside of the business, no matter where they might be located, providing the ability to quickly and easy share, discuss, and collaborate on documents, presentations and ultimately make more informed and timely decisions.
- **Mitel NuPoint Unified Messaging (UM)**, from basic voice mail to advanced unified messaging, Mitel NuPoint Unified Messaging™ (UM) satisfies the diverse messaging needs of your entire user population.
- **MiVoice Border Gateway (MBG)** is a platform for secure communication with remote Tele Workers. It provides secure deployment of multiple network connectivity services in a number of network edge scenarios.

For further information, refer to the product documentation and release notes on *MiCollab (7.x)* available on the *Mitel Knowledge Base*.

Mitel BluStar Server

A presence engine, the Mitel BluStar Server, is an integral part of the MX-ONE 7.x Solution. MX-ONE 7.x provides full integration for line state monitoring using CSTA3 between MX-ONE and BSS.

The Mitel BluStar server insures that directory/contact data and rich presence information, including Blustar Presence, MX-ONE Line state and Calendar integration, is aggregated towards Mitel BluStar Ecosystem clients and terminals as well as towards Mitel BluStar CMG Web 8.0 web users, Mitel InAttend users and FMC users.

For further information, refer to the product documentation and release notes on *BluStar Server* available on the *Mitel Knowledge Base*.

Mitel BluStar for PC

Mitel BluStar for PC delivers high-quality audio and HD video as well as access to a set of UC features from a single client on the users desktop.

Following features are delivered in BluStar for PC:
- SIP SoftPhone
- HD Video communications (peer-to-peer, 4-way if anchored in BluStar 8000i or Blustar for Conference Room
- Professional audio processing (echo cancellation, automatic gain control, codec support)
- CTI Handphone control
- Instant Messaging
- Desktop Sharing
- Contact integration (BSS, AD, LDAP, Personal Outlook Contacts, CMG)
- Session Border Controller (SBC) support enabling remote access
- Microsoft Office Integration Microsoft Lync/Skype-for-Business plug-in integration
- Integrated call control functionality with Jabra and Plantronics headset
- User license handling controlled in MX-ONE
- SRTP/TLS

For further information, refer to the product documentation and release notes on *BluStar for PC* available on the *Mitel Knowledge Base*.

Mitel BluStar for iPad & iPhone

Mitel BluStar for iOS 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 iPad and iPhone.

Following features are delivered in BluStar for iOS:
- SIP SoftPhone

- HD Video communications (peer-to-peer, 4-way if anchored in BluStar 8000i or Blustar for Conference Room

- Professional audio processing (echo cancellation, automatic gain control, codec support)

- Contact integration (BSS, AD, LDAP, Personal Outlook Contacts, CMG)

- Session Border Controller (SBC) support enabling remote access

- User license handling controlled in MX-ONE

- SRTP/TLS

For further information, refer to the product documentation and release notes on *BluStar for iPad & iPhone* available on the *Mitel Knowledge Base*.

Mitel BluStar for Android

Mitel BluStar for Android delivers high-quality audio and HD video as well as access to a set of UC features from a single client on the users smart phone, over high quality data networks. The following features are delivered in BluStar for Android:

- SIP SoftPhone

- HD Video communications (peer-to-peer, 4-way if anchored in BluStar 8000i or BluStar for Conference Room, or alternatively via MiVoice Video phone.

- Professional audio processing (echo cancellation, automatic gain control, codec support).

For further information, refer to the product documentation and release notes on *BluStar for Android* available on the *Mitel Knowledge Base*.

Mitel BluStar for Conference Room

Mitel BluStar for Conference Room is a flexible video conferencing tool that is designed to enhance the way you communicate and collaborate. The solution consists of a dedicated central unit, audio unit (S850i), separate camera (Pan/Zoom/Tilt), wireless keyboard for control, HDMI compatible display module.

Key features and benefits are:
- SIP Phone
- Native HD Video
- Supports up to 4-way video conferencing
- 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.
- SRTP/TLS.

NOTE: When using a BluStar for Conference Room with MX-ONE, a video license is necessary, per user.

For further information, refer to the product documentation and release notes on *BluStar for Conference* available on the *Mitel Knowledge Base*.

Mitel CMG BluStar Web

Mitel CMG is a part of the UC suite for MX-ONE 7.x Solution. The CMG product port-folio provides a mix of communication tools for attendants and office users, including advanced call handling, activity and availability management as well as speech services.

For further information, refer to the product documentation and release notes on *Mitel CMG BluStar Web* available on the *Mitel Knowledge Base*.

Mitel InAttend

The Mitel InAttend 2.5 is a scalable attendant console based on open standards. It is the core application in Mitel's attendant offering and an essential part of the Mitel UC suite.

In addition to advanced call handling, Mitel InAttend offers powerful search options, calendar integration, Microsoft Lync/Skype-for-Business and IBM Lotus Sametime presence integration and all necessary information for efficient call handling.

Mitel InAttend integrates with BluStar Server, i.e. is part of the BluStar collaboration suite.

Mitel InAttend uses the SIP trunk interface for media and call signaling, and the CSTA Phase 3 interface for certain call control services towards the MX-ONE Service Node.

For further information, refer to the product documentation and release notes on *Mitel InAttend* available on the *Mitel Knowledge Base*.

Mitel Dialer

From MX-ONE 6.0 the Mitel Dialer PC client is supported.

The Mitel Dialer is a PC based dialer for making call setups via PC from a predefined SIP telephone. Integrates with MX-ONE via the CSTA phase 3 interface.

For further information, refer to the product documentation and release notes on *Mitel Dialer* available on the *Mitel Knowledge Base*.

Fixed Mobile Conversion (FMC)

The following functions are supported in MX-ONE 7.x (and 6.x):
- Introduction of the FMC Controller.
- Provisioning of FMC sers and controller configuration handled centrally from MX-ONE Provisioning Manager.

- BluStar Server based presence integration enabling the user to read and set status information. Supported by Android and iOS.

For further information, refer to the product documentation and release notes on *Mitel Mobile Client* available on the *Mitel Knowledge Base*.

MiContact Center Enterprise

MiContact Center Enterprise (former Solidus eCare or MiCC Solidus) is a Contact Center application which is integrated with the MX-ONE system, but can also be used with many other call managers.

MiContact Center Enterprise is an all-in-one, adaptive and flexible platform for Unified Communications and Collaboration (UCC), mobility, contact center, business process automation, analytics and reporting, as well as service and database integration. InAttend can also be a contact center agent.

Added functionality in MiContact Center Enterprise (9.x) with MX-ONE 7.x are for example:

- Mobile applications for the agent and supervisor.
- Social media connection for handling Facebook and Twitter messages.
- BluStar Agent enhancements.
- Integration with Applications Connectivity Server (ACS) for switch independence

For further information, refer to the product documentation and release notes on *MITEL MICONTACT CENTER ENTERPRISE* available on the *Mitel Knowledge Base*.

Interfaces and Protocols

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

Figure 1:   Protocols - high level view
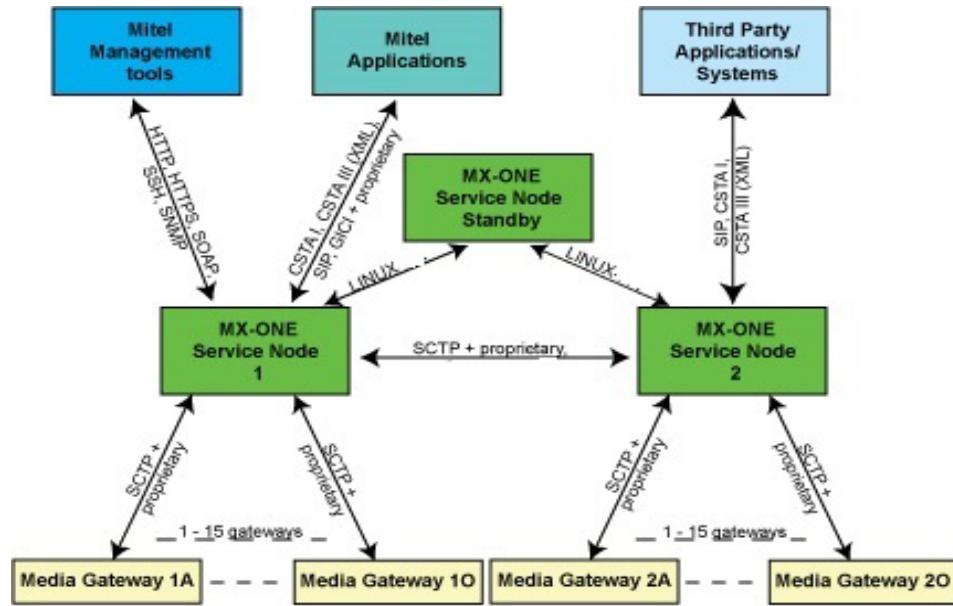
Figure 2:   Protocols - detail level view
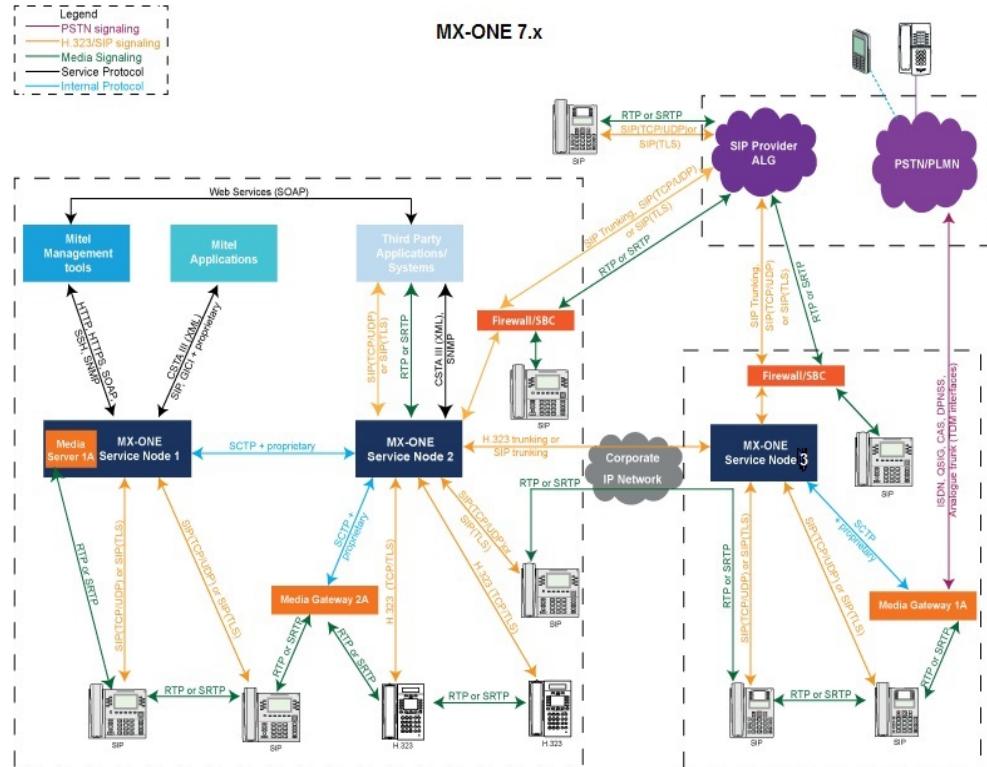
Figure 3:   Interfaces and protocols in MX-ONE and MX-ONE Manager
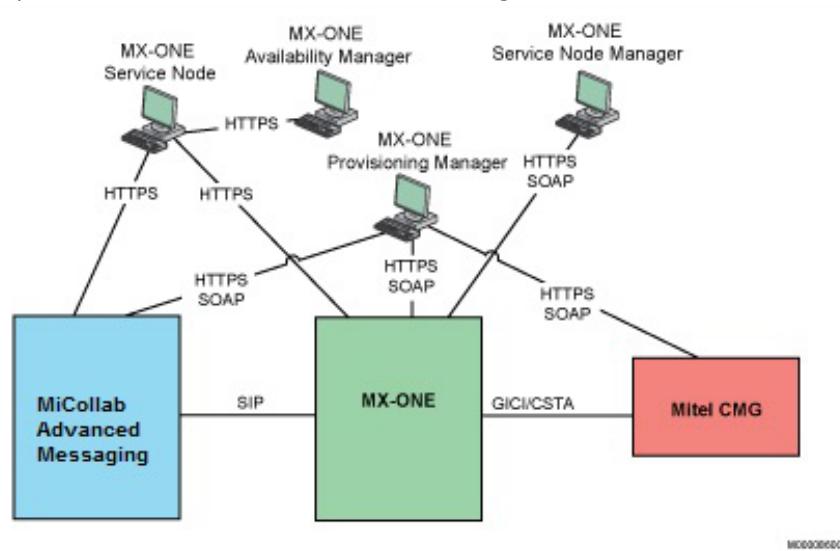
Figure 4:   Interfaces and protocols in MX-ONE
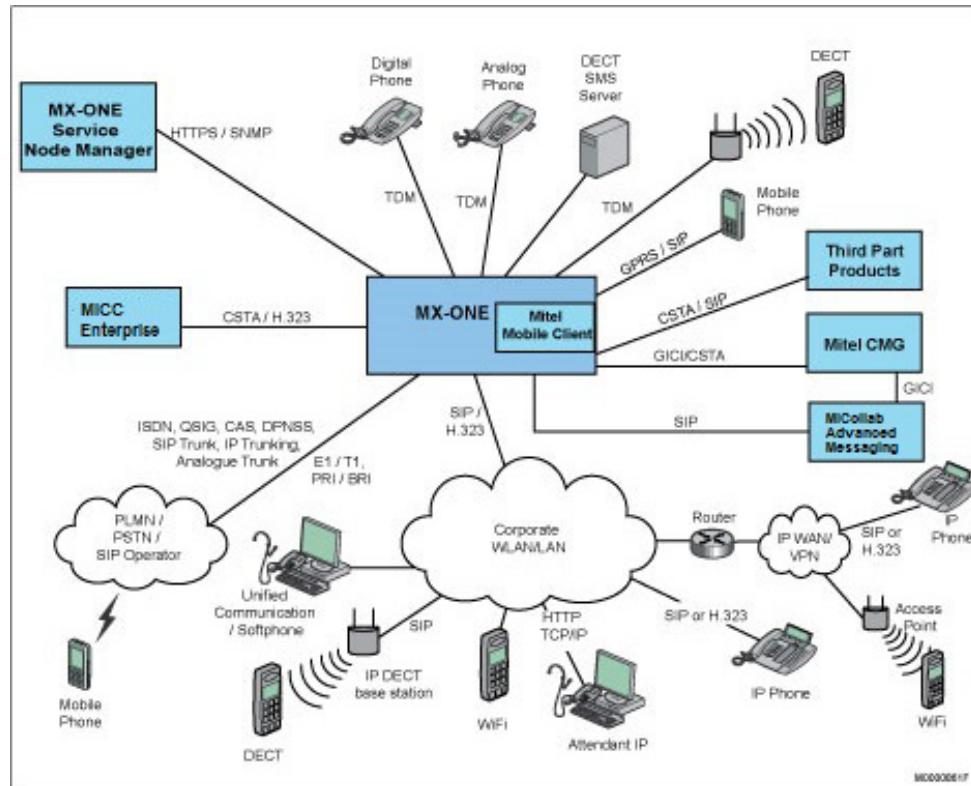
The following interfaces and protocols are available for MX-ONE:

• Private networking by means of SIP tie-line.

• Private networking by means of H.323 tie-line (transporting ISDN QSIG or proprietary ISDN signaling).

• 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 SIP trunk (with various network operator profiles)
- Public ISDN over E1 (30B+D), T1 (23B+D) and BRI (2B+D)
- H.323 extension
- SIP extension (also used for IP DECT, SIP DECT, WiFi, Terminal Adapter and MiCollab Advanced Messaging client)
- SNMP
- Secure Shell (SSH v.2)
- HTTP(S)
- Interfaces to an accounting system (for CIL data)
- DECT (TDM, integrated)
- DASL (DTS)

The following interfaces and protocols are available for CTI:
- TSAPI (Novell)
- CSTA Phase 3 with XML and WebServices
- TR87 (CSTA Phase 3 XML via SIP)
- 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 MiCollab Advanced Messaging:
- SIP
- H.323
- SNMP
- VSI

- Internet Message Access Protocol (IMAP) v4
- Messaging Applications Programming Interface (MAPI)

The following interfaces and protocols are available for MX-ONE Provisioning Manager:

- Web Services (SOAP)
- XML
- HTTP
- HTTPS

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

Migration

MX-ONE supports migration of MD110/Telephony Switch to a MiVoice MX-ONE 7.x system. The major steps of the migration procedure that uses MX-ONE Classic magazines include:

1) Install the servers needed after migration.

2) Install MX-ONE version 7.x and configure the system.

3) Make a version 7.x backup of each server.

4) Stop the MD110/Telephony Switch and replace a limited number of boards.

5) Connect the media gateways to the servers.

6) Start the MX-ONE 7.x.

For detailed information, see installation instructions for *MIGRATING MITEL TSW/MD110 TO MX-ONE 7.1*.

MX-ONE supports Smooth migration. The Smooth Migration can be used for customers who want to migrate a large system in stages (or 'merge' a network of systems) and at the same time do a complete hardware refresh, e.g. when moving from MITEL TSW/MD110(s) to a new MX-ONE.

For detailed information, see operational directions for *Smooth Migration*.

In MX-ONE 7.x there is functionality that makes it possible to migrate TDM extensions to SIP or Mobile Extensions by commands. This is initiated by purchasing appropriate Extension Migration licenses.

Operation and Maintenance

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.

Fault Management

MX-ONE provides basic fault management.

MiVoice MX-ONE

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. Alarms in MX-ONE 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 Service Node has an alarm, device status event and emergency call handling function. An SNMP-interface exposes standard MIB II information and alarms generated in the Service Node can be sent as SNMP traps. A separate optional MIB for Emergency calls can generate an event and alarm when an emergency call is made.

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.

CMG

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

MiCollab Advanced Messaging

The MiCollab Advanced Messaging 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.

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 Service Node Manager 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 Provisioning Manager and comprises data that is used by two or more components, for example user data (used by MX-ONE Provisioning Manager and CMG), extension data (used by all components), and department data (used by MX-ONE Provisioning Manager and CMG).

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

MX-ONE Service Node Manager

MX-ONE Service Node Manager is used to manage telephony system data, for example:

- MX-ONE Service Node Server configurations
- Number plans
- Common service profiles
- Groups
- Routes and trunks
- Media Gateways
- IP/SIP phone and client provisioning

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

For detailed information, see the user guide for *MX-ONE SERVICE NODE MANAGER*.

MX-ONE Provisioning Manager

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

- Adding, changing and removing users

  When managing user data in Provisioning Manager the corresponding user data is automatically updated in the CMG SIP DECT and MX-ONE databases. User account administration, e.g. unlocking users. Including certain group member administration.

- Adding, changing and removing extensions

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

- Adding, changing and removing administrators with multiple access levels, security profiles, service groups, departments, locations, and subsystems.

- Managing mail box services

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

- Managing MiCollab users, FMC users and SIP DECT user services

When managing these services in Provisioning Manager, the corresponding data is automatically updated in the MiCollab Advanced Messaging 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 Provisioning Manager in an existing MX-ONE installation.

• Exporting user and extension data

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

• End User self-service
• Multi Tenanting

Multi Tenanting (customer groups) can be handled, with extension feature levels, and also different management authority levels.

Active Directory and Mapping of User-Defined Fields for PM

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

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

With this release, it is possible to configure MX-ONE Provisioning Manager 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. Other news in this release is AD for Log-on and failure feed back.

Mitel Performance Analytics

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

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 MX-ONE Provisioning Manager.)
• 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.

For further information, refer to the product documentation and release notes on *CMG Server* available on the *Knowledge Base*.

MiCollab Advanced Messaging - Unified Messaging

The following are some examples of data that is configurable for MiCollab Advanced Messaging 7.0 Fax Mail:

- Settings for fax support on the Service Node
- 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 administrator's guide for *Captaris RightFax*.

The following are some examples of data that is configurable for MiCollab Advanced Messaging 7.0 Voice Mail:

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

With the release of MX-ONE 5.0 SP4, integration with MiCollab Advanced Messaging (former OneBox) 5.1 was introduced. This version is a dot release package and brings some enhancements primarily in the MiCollab Advanced Messaging mobile client side. Please refer to the MiCollab Advanced Messaging CPI documentation for a list of enhancements.

There will be continued support for the existing MiCollab Advanced Messaging SP3 SU3, which is a correction package for MiCollab Advanced Messaging 5.0 SP3. There are no new features included with this correction package.

Customers with Software Maintenance - Assurance have the possibility to upgrade their MiCollab Advanced Messaging (former OneBox) 5.0 SP3 system to the MiCollab Advanced Messaging 5.1 version. A new license key will be needed when upgrading to the MiCollab Advanced Messaging 5.1 version.

For further information, refer to the product documentation and release notes on *Mitel MiCollab Advanced Messaging* available on the *Knowledge Base*.

SIP DECT

Configuration of the SIP DECT system and radio network is made via the Open Mobility Manager (OMM) that is part of the SIP DECT package.

The following are some examples of data that is configurable for SIP DECT:

- Connections to MX-ONE IP extensions
- Redundancy and load-share configurations
- Configurations of RFP's in the system
- Synchronization settings for the DECT air synchronization
- Configuration of Locating sunction

For detailed information, see the description for *SIP DECT*.

BluStar Server

The BluStar Server (BSS) is a common application server for all BluStar end-points. The following are some examples of data that is configurable for BSS:

- Configuration of CSTA link to MX-ONE
- Configuration of connection to calendar system and to directory system
- Presence handling
- Integration for presence federation with Lync/Skype-for-Business

For detailed information, see the description for *BLUSTAR SERVER*.

Accounting Management

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

MX-ONE 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 Service Node 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*.

Quality of Service

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

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

Performance Management

MX-ONE provides the following level of Performance Management:
- Basic Performance Management (always included)

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 *MiVoice MX-ONE* and *MiCollab Advanced Messaging*.

MiVoice MX-ONE

MX-ONE provides the following monitoring and performance management functions:
- Congestion on routes

  The rate of congestion on routes is a critical resource in the Service Node 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 Service Node. 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).

MiCollab Advanced Messaging

The following three different tools are available for configuration in MiCollab Advanced Messaging:
- Support Messaging System Management (a suite of Windows-based tools)
- Service Node Diagnostics
- Line Status & Reports

NOTE: *The Service Node Diagnostics tool does not specifically refer to the component Service Node in MX-ONE and the tool is not part of MX-ONE.*

Security Management

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

User Management

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

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.

Protection of VoIP

MX-ONE supports the use of secure real-time transport protocol (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 Service Nodes.

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:

*   Secure RTP (SRTP) to protect media streams

    MX-ONE supports the use of SRTP for media encryption in the IP phones and in MGU/MS 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.

The servers are MX-ONE running 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 anti-virus software updates. It is recommended to use some type of anti-virus software and to have automatic updates, of the security patches, activated. To overcome the VLAN separation, server farms should be protected by fire walls 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.

From MX-ONE 6.0 the VoIP security keys have been modified to be compliant with the EU legislation 388/2012 (former 1232/2011). A VoIP security key is only allowed to work in one installation.

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

Capacity

MiVoice MX-ONE

For information about the capacity and minimum environment requirements of MX-ONE configurations, see the description for *CAPACITIES (for devices) and MX-ONE FEATURE MATRIX (for features/services)*.

The minimum environment requirements differ for virtualized and non-virtualized environments.

For more information about MX-ONE in a virtualized environment, see *MX-ONE VIRTUALIZATION DESCRIPTION*.

Mitel MiCollab portfolio

For information on MiCollab UCC capacities, refer to the product documentation on *MICOLLAB*.

Mitel BluStar Server

For information on BluStar Server capacities, refer to the description for *BLUESTAR SERVER*, and the product documentation and release notes on CMG BluStar Server available on the *Knowledge Base*.

Mitel CMG BluStar Web

For information on CMG capacities, refer to the product documentation and release notes on CMG available on the Knowledge Base and Info Channel.

Mitel InAttend

For information on Mitel InAttend capacities, refer to the product documentation and release notes on Mitel InAttend available on the Knowledge Base and Info Channel.

MiCollab Advanced Messaging

For information on MiCollab Advanced Messaging capacities, refer to the product documentation and release notes on MiCollab Advanced Messaging available on the Knowledge Base.

SIP DECT

For information on SIP DECT capacities, see *the description for SIP DECT*.

MiContact Center Enterprise

For information on MiContact Center Enterprise capacities, refer to the product documentation and release notes on MiContact Center Enterprise available on the Knowledge Base.

Environmental Conditions

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.

For information on climatic environment requirements, see *MX-ONE SITE PLANNING ENVIRONMENTAL SPECIFICATION*.

EMC, Safety and Telecom Compliance

For Aastra/Mitel branded products, when needed, contact your nearest Mitel office for support

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For safety related information, please read *SAFETY INFORMATION*.

For other products delivered with Aastra/Mitel branded products, when needed, refer to documentation delivered with the products or contact manufacturers for details.

References

The following IETF RFCs are supported by the MX-ONE. E = Extension interface, T = Trunk interfaceE = SIP Extension interface, T = SIP Trunk interface, - = other interface. E = Extension interface, T = Trunk interface

RFC	Description	E	T
2246	Transport Layer Security (TLS)	x	x
(2327)	SDP; Session Description Protocol (replaced by 4566)	x	x
2617	HTTP Authentication. Basic and Digest Access Authentication	x	x
2782	A DNS RR for specifying the location of services (DNS SRV)	x	x
(2833)	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals (Replaced by 4733)	x	x
2976	The SIP INFO method	x	x
3003	The Audio/MPEG Media Type	-	-
3261	SIP, Session Initiation Protocol	x	x
3262	Reliability of Provisional Responses in SIP (PRACK method)	x	x
3263	Locating SIP Servers	x	x
3264	Offer/answer method (SDP usage)	x	x
3265	SIP specific event notification	x	x
3280	Internet X.509 Public Key Infrastructure CRL Profile	x	x
3310	Digest Authentication	x	x
3311	SIP UPDATE method	x	x
3323	Privacy mechanism for SIP		x
3325	Private Extensions to SIP for Asserted Identity within Trusted Networks		x
3326	The Reason header field for SIP	x	x
3398	ISDN User Part (ISUP) to SIP mapping	x	x
3407	SDP Simple Capability Description	x	x
3420	Internet Media Type Message/sipfrag	x	x
3428	Extension for Instant Message method	x	

3515	SIP Refer method	x	x
3550	Real Time Transport Protocol	x	x
3551	RTP Profiles for Audio and Video Conferences with minimal control	x	x
3581	Symmetric response routing		x
3665	SIP Basic Call Flow Examples	x	x
3680	SIP Event package for Registration	x	
3711	The Secure Real-time Transport Protocol (SRTP)	x	x
3725	Third party call control in SIP	x	
3824	Using E.164 numbers with Session Initiation Protocol		x
3842	Message waiting indication event package for SIP	x	
3891	The SIP 'Replaces' header	x	x
3892	SIP Referred-by mechanism	x	x
3911	The SIP Join header		x
3960	Describes how to manage early media in SIP using two models: the gateway model and the application server model. The Service Node supports the gateway model.	x	x
3966	The Tel URI for telephone numbers		x
3986	Uniform Resource Identifiers (URIs)	x	x
4028	Session timers in SIP	x	x
4040	RTP Payload Format for 64 kbit/s Transparent Calls	x	x
4235	Dialog event package	x	x
4244	An extension to SIP for Request History Information		x
4320	Actions addressing identified issues with the SIP non-Invite transaction	x	
4346	Transport Layered Security (TLS) 1.1 and SRTP	x	x
4497	Interworking between SIP and ISDN QSIG	x	x
4566	Session Description Protocol (replacing RFC 2327)	x	x
4568	SDP Security Descriptions for media streams (SDES)	x	x
4733	RTP payload format for DTMF digits, Telephony Tones and Telephony Signals (Replacing RFC 2833).	x	x
4904	Representing Trunk groups in tel/sip URIs		x

4975	The Message Session Relay Protocol (MSRP)	x	x
5022	Media Server Control Markup Language and Protocol	x	x
5116	An interface and algorithms for Authenticated Encryption (AEAD)	x	x
5168	XML schema for media control	x	
5246	Transport Layered Security (TLS) version 1.2	x	x
5282	Using Authenticated Encryption Algorithms with the Encrypted Payload of the Internet Key Exchange version 2 (IKEv2) Protocol (AES-128/256 in GCM mode in secure RTP)	x	x
5334	Ogg encapsulation, interface for external streaming sources	x	
5626	Managing Client-initiated connections in SIP	x	
5627	Obtaining and using globally routable User Agent URIs (GRUU) in the SIP	x	
5806	Diversion information in SIP	x	x
5850	A call control and multi-party usage framework for the SIP		x
6116	The E.164 number to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)		x
6140	Registration of Multiple Phone Numbers in SIP		x
6188	The use of AES-192 and AES-256 in secure RTP	x	x
6337	Usage of the Offer/Answer Model	x	x
6910	Call Completion for SIP		x
7315	Private Header Extensions for SIP for the 3GPP (PANI for Emergency calls)		x

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

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

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