

CPI News in MiVoice MX-ONE 6.3

PRODUCT REVISION INFO



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GENERAL

This document describes changes in the MiVoice MX-ONE documentation due to new and changed functionality in MiVoice MX-ONE 6.3 and 6.3 SPx.

For detailed information on the MX-ONE 6.3 Solution, see *MiVoice MX-ONE Solution Overview*, *MiVoice MX-ONE System Description* and other CPI documents.

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CPI LIBRARY NEWS

2.1

MITEL BRANDING AND NAMES

2.1.1

BRANDING

Note! Some documents contain old names and brand, for example name of configuration files and links. These will be phased out over time.

2.1.2

NEW NAMES

Table 1

New product/solution naming	Previous product/solution naming
MiVoice MX-ONE	Aastra MX-ONE
MiVoice MX-ONE	MX-ONE TS
MX-ONE Service Node	MX-ONE TSE
MX-ONE Service Node Manager	MX-ONE MTS
MX-ONE Provisioning Manger	MX-ONE MP
MX-ONE Traffic Manager (MTM)	MX-ONE MSP
Mitel Performance Analytics (former MarWatch, replacing MA)	MX-ONE MA
Mitel TSW (phased out)	TSW (phased out)
MiCollab Advanced Messaging	Mitel OneBox
MiContact Center Enterprise	MiCC Solidus, Solidus eCare
Microsoft Skype for Business	Microsoft Lync
Mitel MX-ONE Chassis	Aastra MX-ONE Chassis
Mitel Server Unit	Aastra Server Unit

New product/solution naming	Previous product/solution naming
Mitel ASU	Aastra ASU
Mitel 69xx SIP Phone	--
Mitel 68xx SIP Phone	Aastra 68xxi SIP Phone
Mitel 67xx SIP Phone	Aastra 67xxi SIP Phone
MiVoice 4200	Aastra 4200
MiVoice 4400	Aastra 4400
Mitel 7100	Aastra 7100
Mitel 1023	Aastra 1023i
Mitel TA7100	Aastra TA7100
Mitel DTxxx	Aastra DTxxx

3 CPI NEWS IN MIVOICE MX-ONE 6.3

3.1 NEWS AND CHANGES IN DOCUMENTS, 6.3 SP7

3.1.1 ANONYMOUS TEXT REMOVAL FOR INCOMING CALLS WITHOUT CLI

If the calling line identity is restricted (CLIR valid) or missing, the display information sent to a called SIP extension has an option that can control what data must be sent to the terminal, either an "anonymous" text string (according to RFC 3261), or another text string configured in the SIP extension default file,

For more information, see the following documents:

- MiVoice MX-ONE IP Extension - Operational Directions
- MiVoice MX-ONE SIP Extension - Description
- MiVoice MX-ONE Administrator Guide - Operational Directions

3.1.2 ENHANCEMENTS TO INDIVIDUAL CALL PICK UP BLOCKING

An enhancement to the Individual Call Pick up Blocking feature enables users to block/restrict the service for certain (or most) extensions. An extension can now answer a call ringing on another extension park a call, or pick up the call from another extension using the same procedure.

Call back calls and calls where both the calling party and answering party have parked calls, cannot be picked up. The Individual Call Pickup service is generally available to all extensions, but can be turned off for the entire system (I/O command, AS, PARNUM

20) by setting no suffix digit (value #F/15), which will also block service requests through key, menu, and CSTA3 service request for relevant extension types.

The enhanced service can be used to request individual Call Pick up for a busy extension. The directory number for which this service is requested is included in the request. This service is supported only for the XML protocol.

For more information about individual call pick-up enhancements, see the following documents:

- MiVoice MX-ONE Technical Reference Guide, MML Parameters - Parameter Description

3.1.3

LICENSING OF THIRD-PARTY DECT PHONES

New licensing of third-party DECT phones (different from Mitel DECT phones) will be supported. Users can use the same license that are used for third-party SIP phones, already renamed as third-party devices. Integrated DECT phones can now be adjusted to be compatible with any required third-party device license agreements for non-MX-ONE devices.

For more information about licensing of third-party DECT phones, see the following documents:

- Upgrading or Updating MX-ONE 7.X-Installation Instructions
- Cordless Phone - Installation Planning
- Cordless Phone - Description

3.2

NEWS AND CHANGES IN DOCUMENTS, 6.3 SP6

3.2.1

RINGING TIME OUT LIMITATION FOR DTS AS GROUP HUNT MEMBER

When call list contains a PBX group the IRD execution ends, and diversion on no reply is rejected if the group member has a diversion on no reply back to the group (member is a DTS). This results in that if the group member is not answering the call next free member in the group is not selected and call is ended.

Therefore, an enhancement in the GH functionality is being done by restricting the ringing time for extension. In case of group call, the queue time and/or the ringing time specified for the selected group is the timer(s) which will be used and NOT the alerting timer specified in the personal number list for the answering position.

For more information, see the following document:

- MiVoice Personal Number - Description

3.2.2

BLF/MNS-TRANSFER KEY ON MITEL SIP PHONES

A new key type MNS-Xfer (MXFER) is provided for SIP extensions. The key provides both MNS- and transfer functionality. Now the BLF/Xfer combines the BLF key and Xfer key's functionality together that allows user to just press a key once to transfer a call. This is supported on all Mitel 6900 and 6800 SIP phones with MX-ONE.

For more information, see the following document:

- Mitel 6940 SIP Phone for MX-ONE, QUICK REFERENCE GUIDE

- Mitel 6930 SIP Phone for MX-ONE, QUICK REFERENCE GUIDE
- Mitel 6920 SIP Phone for MX-ONE, QUICK REFERENCE GUIDE
- Mitel 6873 SIP Phone for MX-ONE, QUICK REFERENCE GUIDE
- Mitel 6867 SIP Phone for MX-ONE, QUICK REFERENCE GUIDE
- Mitel 6867 SIP Phone for MX-ONE, QUICK REFERENCE GUIDE
- MiVoice MX-ONE Technical Reference Guide, Unix Commands - Command Description

3.2.3 CALL CANNOT BE DEFLECTED BEFORE RVA QUEUE MESSAGE IS PLAYED

Whether an RVA can be interrupted or not when call is queued to ACD group. If the setting is to play complete voice message, then a queued call to a CTI group with RVA queue message cannot be deflected until the RVA is over (played).

For more information, see the following document:

- Recorded Voice Announcement, RVA, Description
- Automatic Call Distribution, AC, Description

3.2.4 REPLACEMENT OF THE DELL R330 SERVER WITH A NEW R440 SERVER FOR MX-ONE

The Dell R330 (87L00053AAA-A) model/server that was being used by MX-ONE and was running with SLES11 or SLES12. Now, the Dell R330 server is going to be replaced by the new Dell R440 server for MX-ONE.

If these ports are not open, then the MiCW fails to connect to server at the start of the wizard.

For more information, see the following document:

- MiVoice MX-ONE Capacity - Description
- MiVoice MX-ONE Hardware Status and Reliability ASP 113 01 - R-STATE SURVEY
- MiVoice MX-ONE Engineering Guidelines
- MiVoice MX-ONE, MiVoice MX-ONE Site Planning
- MiVoice MX-ONE Installing and Configuring - Installation Instructions

3.2.5 THE 68XX TERMINALS DO NOT REGISTER WITH DNS_SRV RECORDS

MX-ONE does not distribute SIP registrations. Phones do not switch to secondary server when primary is out of order. SNM does not support to create config files with DNS_SRV. So, startup.cfg and aastra.cfg has been created manually. Terminal reads the config files correctly and resolves the DNS_SRV and gets the correct server ip addresses, though login fails.

For more information, see the following documents:

- MiVoice MX-ONE Technical Reference Guide, Unix Commands - Command Description

3.2.6

THE CODEC G.723 IS REMOVED FROM THE IP_DOMAIN COMMAND

The G.723 is removed, since there is no supported media gateway that supports codec any more (since 6.0). This codec is used in gateway calls from/to IP terminals and SIP trunks.

For more information, see the following documents:

- MiVoice MX-ONE Technical Reference Guide, Unix Commands - Command Description

3.3

NEWS AND CHANGES IN DOCUMENTS, 6.3 SP5

3.3.1

INTRODUCING MITEL 6970 IP CONFERENCE PHONE

A new phone model 6970 IP Conference Phone is added to the 6900 series SIP Phones. It is an enterprise-level IP conference phone.

The 6970 IP Conference phone can be managed in MX-ONE system. The configuration and setup of 6970 phones in MX-ONE management system is similar to other 6900 phones.

For more information, see the following documents:

- Hardware Status and Reliability
- 6970 SIP Conference Phone, Installation Guide
- Mitel 6970 SIP Phone for MX-ONE, Quick Reference Guide
- 6970 SIP Conference Phone, User Guide
- MiVoice MX-ONE Power Consumption, Description
- Mitel 6900, 6800 & 6700 SIP Terminals for MiVoice MX-ONE, Installation Instructions
- MiVoice MX-ONE Terminal Overview, Description
- MiVoice MX-ONE Extension Functionality Comparison
- MiVoice MX-ONE Feature Matrix

3.3.2

UPDATED PORT LIST USED BY PM/SNM

TCP/IP ports: 10255, 10256, 10257, 10258, 10259, and 10260 are used for provisioning users to MiCollab. All these ports are open on the MSL IP address (third-party). All are external ports on the MiCollab server and provide external Application Programming Interfaces (APIs) with access to the MiCollab system. These APIs can be used to support management applications.

If these ports are not open, then the MiCW fails to connect to server at the start of the wizard.

For more information, see the following document:

- MiVoice MX-ONE System Planning - Description

3.3.3

UNANSWERED WAITING CALLS WERE NOT SHOWN IN INTEGRATED DECT

This describes that if a user is busy on the DECT when someone tries to call this user, after hanging up the first call, the user should be able to see who has been trying to reach him out (For example, doctors at Hospitals). This is done by doing a short “automatic call waiting” to update the missed call list in the DECT terminal.

For more information, see the following documentation:

- MiVoice MX-ONE Technical Reference Guide, Unix Commands - Command Description

3.3.4

DEFLECT OR SST OF CALL IN HELD STATE IS NOT SUPPORTED

This main service in the outbound call context is used to move a call-in speech (single-step transfer) from a monitored extension or ACD queue to a new destination. Deflect or SST of Call is supported only in ringing state, not in held state.

For more information see the following documentation:

- MiVoice MX-ONE Computer Supported Telecommunications Applications (CSTA with AppLink), CS - Description

3.3.5

FLEXIBLE MOH/RVA SETTINGS FOR EXTERNAL AND INTERNAL CALLING PARTIES

The PARNUM=116 is modified for Recorded Voice Announcement (RVA) so that it is possible to set separately for Welcome and/or Continuous (MoH) announcement to be active for individual extensions having different settings depending on the origin of the call (that is, whether the calling party is external/internal).

For more information, see the following documentation:

- MiVoice MX-ONE Technical Reference Guide, MML parameters

3.3.6

LIMITATIONS FOR SCA AND SCABR CONFIGURATION

This describes the number of SCAs can be programmed for each server in each system.

The Mitel 6700/6800/6900 SIP terminal models support this feature. All the supported 15000 SIP terminals can have the SCA and SCABR feature configured.

The SCA and SCABR feature requires a line key. Depending upon the model, there can be up to 40 SCA/SCABR keys. The maximum number of SCA/SCABR keys that can be initiated depends upon the model. SCA/SCABR can have up to 40 members, that is supervising SIP extensions

For more information, see the following documentation:

- MiVoice MX-ONE Shared Call Appearance, description

3.3.7 THE 6867I PHONE DISPLAYS ONLY 30 STREAMING CHANNELS

The `streaming_data` command can be used to configure data for 68xx/69xx terminals to request streaming. The configuration is supported for both terminals and media server. For 68xx/69xx phones, the number of channels that can be initiated and presented per key is limited to 30.

The number of different streamed announcements and streaming sources that can be initiated is not limited, other than by the network bandwidth, the maximum number of SIP extensions, and the number of dedicated feature keys. For Mitel 6800/6900 terminals, the number of MOI channels is limited to 30.

Only IPv4 is supported (because the terminals used do not support IPv6). There is a timer that limits a Streaming/Music on Idle session to 2 hours.

For more information, see the following documentation:

- MiVoice MX-ONE Technical Reference Guide, Unix Commands - Command Description
- MiVoice MX-ONE Streaming on extensions - Description

3.3.8 VOICE MAIL SYSTEM GETS WRONG GICI MESSAGE FROM PBX

When there is an incoming external call to an extension, the external calling number was being truncated by the `call_list` active command and the call, deflected to the voice mail group. Such calls were being reported to the voice mail group with the record "STX 91 D T V CR LF" instead of "STX 96 PUB T V CR LF".

To resolve this error, a correction has been made in the code for "Diversion on no-reply" for calls from public origins in the Voice Mail interworking description document. The MX-ONE now sends VM record with code "STX 96" instead of STX 91 for external (public) calls diverted on answer message.

Also, the STX 04 message, which was missing in the document, has been updated with the following statement:

A request from an external information system can start a PBX-ordered update of message diversion or message waiting missions. This request enables the PBX to remove all message diversion or message waiting missions before updating from the external information system.

For more information, see the following documentation:

- MiVoice MX-ONE Voice Mail, VM - Description

3.3.9 SERVERS PASSWORDS MUST BE CONSISTENT IN MULTI LIM INSTALLATIONS

In the MX-ONE Installation setup (which includes passwords for user accounts `root`, `mxone_admin`, and `mxone_user`), all passwords for different servers of multi-server systems must be identical.

For more information, see the following documentation:

- MiVoice MX-ONE Installing and Configuring - Installation Instructions

3.3.10 LOG MISSED CALL AT BUSY FOR INTEGRATED DECT

An option to log missed call at busy is introduced for Integrated DECT. This is configured by using the new parameter `log-missed-call-at-busy` in `dect_system_id` command.

For more information, see the following documentation:

- Cordless Extension, Operational Directions
- Technical Reference Guide, Unix commands

3.4 NEWS AND CHANGES IN DOCUMENTS, 6.3 SP4

3.4.1 IP PHONE SERVER UPGRADE

The Apache Tomcat version used in IP Phone Server installation package has been upgraded from 8.0 release to 9.0.19 release.

Place in the tree view >> Up and Running/MX-ONE Manager.

3.4.2 SIP TRUNK PROFILE FOR TELENOR SIP TRUNK

A new sip trunk profile for Telenor is added. This describes about the SIP Trunk, configuration, and SIPLP traces. The sip_route command is updated with two more new parameters -accept and -addheader with additional examples such as % sip_route -print -profile and % sip_route -print -profile Lync_TLS.

Place in the tree view >> Up and Running/Operation & Maintenance>> Commands and Parameters.

3.4.3 REMOTE BRANCH NODE INTER-SERVER MODELLING

Inter-lim connection lost - ip packet fragmentation needed.

When low speed WAN links are used ICMP has to be enabled in the network to support fragmentation in case Inter-LIM connection is limited in frame size.

Place in the tree view >>Planning>>MX-ONE System Planning.

3.4.4 AD SYNCHRONIZATION IN PM DOES NOT WORK WITH MORE THEN 5 ACTIVE DOMAINS

There is a limitation in AD, it allows only to have five subscriptions for automatic change notifications. AD synchronization in PM does not work with more than five active domains. If you activate more than five domains (green icon), the automatic synchronization will not work anymore.

Place in the tree view >>Up and Running>>MiVoice MX-ONE Manager/MX-ONE Provisioning Manager.

3.4.5 PAGING SERVICE INITIATED FROM A SIP EXTENSION WITH SUFFIX DIGIT AT RINGING

To initiate a Paging call, you must dial 7(), or for Mitel 6700/6800/6900 SIP phones, press Paging function key.

Place in the tree view >>About MiVoice MX-ONE/MiVoice MX-ONE Feature Matrix.

Place in the tree view >>Planning/Extension Functionality Comparison.

Place in the tree view >>Up and Running>>Feature & Interworking Descriptions.

Place in the tree view >>Up and Running>>Operation & Maintenance>>Operational Directions.

Place in the tree view >> Up and Running/Operation & Maintenance>> Commands and Parameters.

Place in the tree view >> Up and Running/Operation & Maintenance>> Directions for Use.

3.4.6 RVA ENHANCEMENTS

All Media Gateways must have same set of voice prompts.

For proper configuration, all Media Gateways (MGUs and Media Servers) must have the same set of local voice prompts installed. However, If all Service Nodes are configured to use Media Server (SIP or MSCML interface) for RVA and MOH, then only Media Servers need to have voice prompts installed.

Place in the tree view >>Up and Running>>Operation & Maintenance>>Operational Directions.

3.4.7 CONNECTED GUESTS UPDATES

This describes how to configure the integration between the MX-ONE and the Connected Guests. In addition, this also address the common interfaces information such as SIP routes, Voice Mail/MW, CIL and CSTA phase 3, and certain AS-parameters.

Place in the tree view >>Up and Running>>Feature & Interworking Descriptions>>Hospitality Application.

Place in the tree view >>Up and Running>>Operation & Maintenance>>Operational Directions.

3.4.8 A-NUMBER ENHANCEMENT FOR EMERGENCY CALLS

If ISDN is used as the only access to the public network for calls to the emergency center, you will need to configure how to handle location identity (ELIN) information, and there maybe limitations on the possibility to call back to the A-number. The ISDN trunk can be configured (with VARC D1 for TL60) to replace either A-number, A-Name, or both.

Place in the tree view >>Up and Running>>Operation & Maintenance>>Operational Directions.

3.5 NEWS AND CHANGES IN DOCUMENTS, 6.3 SP3

3.5.1 ATTAINING GENERAL DATA PROTECTION REGULATION (GDPR) COMPLIANCE IN MIVoice MX-ONE INCLUDED

The intent of GDPR is to harmonize data privacy laws across Europe so that the data privacy of EU citizens can be ensured. GDPR requires businesses to protect the personal data and privacy of EU citizens for transactions that occur within EU member states. GDPR also addresses the export of personal data outside of the EU.

Place in the tree view >> Up and Running/Operation & Maintenance/Feature and Interworking Descriptions.

3.5.2 CSTA III APPLICATION SESSION AUTHENTICATION SERVICES ADDED

The CSTA Phase 3 interface (XML & TR87 protocols) has added support for the Application Session Authentication services, i.e. the ECMA-354 Standard. The authentication criteria must be configured, so there are new commands (*csta_authentication*) for that.

Place in the tree view >> Up and Running/Operation & Maintenance/Feature and Interworking Descriptions/CSTA.

Place in the tree view >> Up and Running/Operation & Maintenance/Commands & Parameters/Technical Reference Guide, unix commands, Command Description (*csta_authentication*).

Place in the tree view >> Up and Running/Operation & Maintenance/Operational Directions/CSTA/CSTA Server (Phase III) Operational Directions.

3.5.3 MITEL TRAFFIC MANAGER DOCUMENTATION UPDATED TO VERSION 1.4

The Mitel Traffic Manager and its documentation has been updated to version 1.4.

Place in the tree view >> Up and Running/MX-ONE Manager/MX-ONE Traffic Manager/MTM Getting Started, Technical Guide, and User Guides.

3.5.4 AUTHENTICATION DATA ENCRYPTION INFORMATION UPDATED

As per the GDPR compliance, the authentication data encryption information is updated to handle authorization codes.

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (*auth_code*).

3.5.5 FEATURE LEVEL H323 UPDATED

As per the GDPR compliance, the authentication data encryption information is updated to handle authorization codes.

Place in the tree view >> Up and Running/Feature & Interworking Descriptions/License Handling.

3.5.6 CSTA SERVER (PHASE III) UPDATED

Operator and/or call origin group removed from the CSTA Phase III, Description document.

Place in the tree view >> Up and Running/Feature & Interworking Descriptions/CSTA (Computer Supported Telecommunications Applications).

Place in the tree view >> Up and Running/Operational Directions/CSTA (Computer Supported Telecommunications Applications).

3.5.7 -ACCEPT AND -ADDHEADER PARAMETERS WITH EXAMPLES UPDATED UNDER SIP_ROUTE COMMAND

Parameters (-accept and -addheader) with % sip_route -print -profile and % sip_route -print examples updated under the SIP_ROUTE command.

Place in the tree view >> Up and Running/Operation & Maintenance/Commands & Parameters.

3.5.8 HUNT GROUP MEMBER WITH MULTIPLE TERMINALS, RINGING ON ALL TERMINALS ADDED

Hunt Group member with Multiple terminals, ringing on all terminals. So that, forked extensions can all ring.

Place in the tree view >> Up and Running/Feature & Interworking Descriptions.

3.6 NEWS AND CHANGES IN DOCUMENTS, 6.3 SP2

3.6.1 CALLBACK ENHANCEMENT FOR SIP-TIE-LINE CASES

The services Callback on busy/no reply or on not-available user via SIP tie-line had a limitation, since the SIP tie-line did not support the PUBLISH option of RFC 6910, which meant the Callback could not be re-initiated if the originator could not be recalled (or did not answer the recall).

Now the optional SIP:PUBLISH message is supported, so the callback can be re-initiated as a reversed Callback.

Place in the tree view >> About MiVoice MX-ONE/MiVoice MX-ONE Feature List Description (Callback sections).

3.6.2 CALL LOGGING OUTPUT ENHANCEMENT FOR DNIS IN ACD BACKUP GROUP SIP-TIE-LINE NETWORK CASES

When using the DNIS (Dialed Number Information Service) for ACD/CTI groups, and re-directing the calls to ACD/CTI backup groups located in another system, via SIP tie-line networking, the dialed DNIS number could be lost, i.e. not output to CIL.

Now the dialed DNIS number will be stored as Account Code, and conveyed in the network, and thus be included in the CIL output, in both the originally addressed system and in the backup group's system.

Place in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking Descriptions/Call Information Logging and QoS Logging interface description.

3.6.3 MEDIA PORT RANGE MINIMUM CHANGED

When configuring Media Gateways a minimum of 1000 consecutive media port numbers were required for a media gateway interface. This has been reduced to 200 ports minimum. (For smaller systems and when firewalls restrict the number of available ports). (NFR request.)

Place in the tree view >> Planning/MiVoice MX-ONE System Planning description and Up and Running/Operation & Maintenance/Commands and Parameters/Technical Reference Guide, unix command description (*media_gateway_interface* command)

3.6.4 CALL PROGRESS MESSAGE CHANGE FOR SWITZERLAND

For calls to not-available (logged-off or out-of-range) generic extensions, for example DECT or IP phones, the progress tone message for the caller has been changed back to “busy tone” (from ring tone) for the Swiss market/application system. This was changed in 2012, in MX-ONE 5.0 SP2, but is now reverted, to be consistent with most other markets. (NFR request).

Place in the tree view >> Planning/MiVoice MX-ONE Market Characteristics description (Tone number 3 Busy used in stead of tone 15 Ringing).

3.6.5 NEW CSTA3 XSD INTERFACE DESCRIPTION DOCUMENT

A new Interworking Description document for the CSTA Phase 3 interface has been added, in XSD format, describing the CSTA Services (both standard and proprietary services) supported by the MiVoice MX-ONE Service Node. May be useful for designers of CSTA applications. It provides additional details to the existing CSTA Phase 3 IWD, 3/15519-ANF 901 43 Uen.

Place in the tree view >> Up and Running/Operation & Maintenance/Feature & Interworking Descriptions/CSTA (Computer Supported Telecommunication Applications)/CSTA Phase 3 API in XSD format.

3.6.6 PM AND SNM ENHANCEMENTS

PM and SNM have been modified for security enhancement reasons, and for adaptations to O&M changes. TBD

Place in the tree view >> Up and Running/MiVoice MX-ONE Manager/PM and SNM descriptions.

3.7 NEWS AND CHANGES IN DOCUMENTS, 6.3 SP1

3.7.1 SUPPORT FOR MITEL 6900 SIP TERMINALS

Mitel 6900 SIP terminals are now supported, specifically the Mitel 6920, 6930 and 6940 models, with accessories, in the MiVoice MX-ONE Service Node, and in PM/SNM.

Place in the tree view >> Installing and using/Installing and Administrator's guides, and Installing and using/User Guides and QRGs/Mitel 6900 phone documents.

3.7.2 HUNT GROUP MEMBER STATUS DISPLAY ON SIP PHONES

The MiVoice MX-ONE now supports display of hunt group member status on Mitel 6800/6900 (and to a limited extent also 6700) SIP phones.

The function is a new monitoring key type, GMA key, and one such key must be assigned per group where the extension is member. The function must also be activated (by new command *extension_group_system*).

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (extension_key modified with new GMA key, and new extension_group_system activation command added).

3.7.3 MULTIPLE DIVERSION ENHANCEMENTS

The Follow-me and Direct Diversion services have been enhanced to allow up to 5 consecutive hops within one single system. (Earlier it was usually maximum 2 hops).

The function means a modification of the diversion logic for all extension and extension group types, allowing more hops for the Follow-me and Direct Diversion services. The other Diversion services are basically not modified.

Place in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking Descriptions/Call Diversion description.

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (diversion_system, modified with a new parameter

3.7.4 MODIFIED EXTERNAL FOLLOW-ME (ECF) INTERACTIONS WITH DIVERTED GROUP CALLS

Some multiple diversion enhancements of the ECF function has been done for specific use cases, such as hunt group calls which are diverted to a DTS, where the DTS has ECF active. The ECF shall be allowed to execute.

Another case is calls to attendant (group), which are diverted (by Night Service Diversion) to an extension which is night service position. The extension should be able to have ECF or Follow-me active and executed.

Place in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking Descriptions/Call Diversion description.

3.7.5 NEW LOCK, MOBILE AND TRANSFER/SPEED-DIAL KEYS ON SIP PHONES

Three new key types are introduced for Mitel 6800/6900 SIP terminals, namely a LOCK, a MOBILE and a TRANSFER/SPEED-DIAL key. Whether the keys can actually be configured may depend on the model. E.g. Mobile key is only for Mitel 6930 and 6940.

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (extension_key).

Place in the tree view >> Installing And Using/User Guides and QRGs/IP/SIP Telephones/Mitel 6800 and 6900.

3.7.6 SNM AND PM ENHANCEMENTS

WebService support is added in Provisioning Manager to allow a user with a DTS phone along with a SIP client (MiCollab Client Softphone).

SNM has added support for the configuration of streaming via Media Server (equal to the *media_server_message* command).

Adaptations for changes in the MX-ONE 6.3 SP1 Service Node O&M functions are also done; for example, new SIP phone key types.

Place in the tree view >> Up And Running/MX-ONE Manager/Service Node Manager & Provisioning Manager.

3.7.7 SECURITY ENHANCEMENTS FOR MANAGEMENT APPLICATIONS

An upgrade has been done to Apache Struts version 1.3.10, to mitigate certain vulnerability issues for management applications.

Cross Site Request Forgery (CSRF) and Server Side validation has also been introduced for PM/SNM for security reasons.

Place in the tree view >> Up and Running/MX-ONE Manager.

3.7.8 BACKUP SCHEDULING ENHANCEMENTS

PM has added support to schedule "Config mirror" jobs.

Place in the tree view >> Getting Started/New Installation & Upgrade.

3.8 NEWS AND CHANGES IN DOCUMENTS, 6.3 (SP0)

3.8.1 PROCEDURES FOR FASTER AND EASIER UPGRADE

Certain enhancements have been done in the upgrade procedures, in order to reduce the system down-time at upgrade. There is a new operational directions document, in addition to the existing ones.

Placed in the tree view >> Getting Started/Upgrade/MiVoice MX-ONE Upgrade Process Minimizing System Down-time.

3.8.2 REDUCED DOWN-TIME FOR REDUNDANCY FAIL-OVER

Enhancements have been done for cases where Server Redundancy is used, utilizing pre-loading to reduce the system down-time.

Placed in the tree view >> Up and Running/Operation & Management/Operational Directions/Server Redundancy.

3.8.3 ENHANCED SECURITY

The SRTP encryption suites AES-256 are supported by the Media Server, MGU and MX-ONE Service Node. It is not yet supported by any Mitel terminals.

Padlock display (indicating encryption of media) on Mitel 6800 SIP terminals has been enhanced, so it can be controlled from the Service Node for gateway use cases.

PIN/Password reset function for MiCollab has been added.

IP Phone Software Server, IPP, now supports HTTPS.

Placed in the tree view >> Planning/Security Description and Security Guidelines Operational Directions. Also MiVoice MX-ONE System Planning.

Placed in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking descriptions/MGU and MGU2 descriptions, and Media Server description.

3.8.4 EMERGENCY CALL ENHANCEMENTS

SNMP event and alarm

Whenever making an emergency call, i.e. dialing a 911/112 number, a new optional SNMP trap and alarm is generated. This trap and alarm can be used by entities like the Mitel Mass Notification application, to provide enhanced Emergency Notification functionality, or by MPA to take some measures.

Multiple ring-back numbers

Another enhancement for emergency calls is the support for multiple ring-back numbers per IP domain.

Place in the tree view >> Up and Running/Operation and Maintenance/Operational Directions MiVoice MX-ONE SNMP Support, Alarm Notification and Emergency Call Events.

Place in the tree view >> Up and Running/Fault handling/MiVoice MX-ONE Fault location and Fault Codes (New alarm 2:11).

3.8.5 STREAMING ON IDLE SIP EXTENSIONS (E.G. MUSIC ON IDLE)

The MiVoice MX-ONE now supports streaming on an idle SIP terminal (Mitel 6800/6900 terminals), which have a dedicated key and menu for the function. This function may be relevant for example for Hospitality solutions.

A couple of new alarms have also been created, 5:29 and 5:30, related to the required Media Server.

Place in the tree view >> Up and Running/Operation and Maintenance/Operational Directions/Streaming on idle extension. (New)

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (extension_key, media_server, streaming_data). (Modified)

Place in the tree view >> Up and Running/Fault Handling/Fault Location, and Fault Codes. (Modified)

3.8.6 STREAMING IN ACTIVE CALLS, FOR RVA, MOH, VOCAL GUIDANCE

The MiVoice MX-ONE now supports streaming for Recorded Voice Announcement, Music on Hold/Wait and Vocal Guidance cases, i.e. as voice progress messages in active calls.

The streaming function requires Media Server(s) configured specifically to support a SIP/Media Server Control Markup Language interface and protocol.

Place in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking Descriptions/Streaming on extension. (New)

Place in the tree view >> Up and Running/Operation and Maintenance/Operational Directions/Streaming Media Server Configuration. (New)

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (media_gateway_config modified, and new media_server and media_server_message commands).

3.8.7 PRESENTATION RESTRICTION ENHANCEMENTS, SEPARATING INTERNAL AND EXTERNAL CALLS

The MiVoice MX-ONE now supports a more flexible handling of presentation restriction for numbers and names, which can be differently set per extension, for internal and external calls. (Could be separately handled before, but only per route, not per extension).

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (extension_profile).

3.8.8 SET FORWARDING FOR AN EXTENSION THAT DOES NOT ALLOW IT, I.E. OVERRIDE WHEN REQUESTED BY INATTEND

The MiVoice MX-ONE now supports a request for override when the served user has prohibition of Set Forwarding, i.e. when Set Forwarding is requested via CSTA phase 3, and the served user extension does not allow it,

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, unix commands (extension_profile).

3.8.9 SERVICE NODE MANAGER AND PROVISIONING MANAGER UPDATES

The SNM and PM applications have been upgraded to latest revision, to comply with management changes in the Service Node. There are new functions for example for media streaming. See the respective documentation for those applications.

Place in the tree view >> Up and Running/Management Applications/SNM and PM descriptions.

3.8.10 PREPARED SUPPORT FOR MITEL 6900 SIP TERMINALS

Support for Mitel 6900 SIP terminals, specifically the Mitel 6920, 6930 and 6940 models, has been added/prepared, in PM/SNM and in the MiVoice MX-ONE Service Node. (However, the terminals are yet to be released).

Place in the tree view >> Installing and using/Installing and Administrator's guides, and Installing and using/User Guides and QRGs/Mitel 6900 phone documents.

3.8.11 MANAGEMENT HYGIENE, COMMAND REWORK

Certain old MML commands have been replaced by new unix style commands. This concerns the *CPXXX*, *FTXXX*, *ROELX*, *SUXXX*, *SYXXX*, and *TCMAX* commands, which have been replaced by new/modified unix style commands, named *resource_status* (expanded functionality), *traffic_matrix*, *global_traffic_data*, *route_data_common*, *function_test* and *vacant_number*.

Some functional changes have also been done (for *resource_status*), primarily to enhance the data collection for the MPA (Mitel Performance Analytics) application.

Place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, MML commands (and parameters) and Technical Reference Guide, unix commands, and related Operational Directions.

3.8.12 PHASE-OUT OF THE DUAL ACCESS FUNCTION (FOR ISDN S0/DTS)

The Dual Access function for ISDN S0 terminals and DTS (sharing the same number) has been removed, i.e. is no longer supported.

This also means alarm/fault code 0:343 has been removed.

Old place in the tree view >> Up and Running/Operation and Maintenance/Technical Reference Guide, MML commands and ditto parameters. (Removed ACCTYP parameter and D11 in the ADC parameter of the KSXXX commands.)

Old place in the tree view >> Up and Running/Operation and Maintenance/Fault Handling/Fault Codes Fault Tracing Directions & MiVoice MX-ONE Fault Location.

3.8.13

SIP EXTENSION VISITOR DESK PHONE (VDP) ENHANCEMENT

A new command vdp_data has been added in order to be able to print and erase VDP data, e.g. key configuration data

Place in the tree view >> Up and Running/Operation and Maintenance/Commands and Parameters/Technical Reference Guide, unix command description (vdp_data command).

Place in the tree view >> Up and Running/Operation and Maintenance/Operational Directions/IP extension (removal).

3.8.14

CSTA PHASE 3 DOCUMENTATION ENHANCEMENT

A new IWD document describing the CSTA phase 3 services and events support (XML protocol PICS proforma information) has been added.

Place in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking descriptions/CSTA Phase 3 description.

3.8.15

CAPACITY ENHANCEMENTS

HLR fail-over can now support all 15000 IP extensions in a server (up from old maximum 3500 extensions).

Least Cost Routing DNT table capacity has been increased from 5000 to 35000 external destinations.

Place in the tree view >> Up and Running/Operation and Maintenance/Feature and Interworking descriptions/HLR backup description.

Place in the tree view >> About MiVoice MX-ONE/MiVoice MX-ONE Feature Matrix description.

3.8.16

APPLICATIONS UPDATED TO LATEST VERSIONS

Optional applications, like MPA, MiCC Enterprise, MiCollab, Mitel CMG and InAttend have been upgraded to latest revision. See the respective documentation for those applications.

Place in the tree view >> Getting started/Optional Installation/MPA, MiCC Enterprise, MiCollab, Mitel CMG.

Several of the applications have separate document libraries.