MiVoice MX-ONE

Feature List

DESCRIPTION

1 INTRODUCTION

This document gives a short description of all features available in the MiVoice MX-ONE. Most of the features described in this document are covered by a basic license, but there are some features that require specific additional licenses.

For information on system and feature capacities, see the description for *CAPACITIES* and see the description for *MIVOICE MX-ONE FEATURE MATRIX*.

For information on terminal types, see the description for *TERMINAL OVERVIEW*.

2 GLOSSARY

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

3 FEATURES

3.1 FEATURE DESCRIPTIONS

For each feature, the following information is presented:

- Feature type
- Definition
- Settings
- Procedure and function

For information on capacity, see the description for *CAPACITIES*.

In this document, features are listed in alphabetical order in a tabular format.

3.2 FEATURE TYPES

3.2.1 SYSTEM FEATURES

System features are features that are fixed or can be programmed in the MX-ONE. These features cannot normally be affected by any type of user terminal. System features can only be used within the MX-ONE or over a network to other connected MX-ONEs or to other Private Branch Exchanges (PBXes).

For information on which user system that can be initiated or terminated for each type of terminal, see the description for *MIVOICE MX-ONE FEATURE MATRIX*.

3.2.2 USER FEATURES

User features can in general be accessed from any user terminal that is connected to the MX-ONE.

For information on which user features that can be initiated or terminated for each type of terminal, see the description for *MIVOICE MX-ONE FEATURE MATRIX*.

3.2.3 NETWORK FEATURES

Network features are generally available only when the MX-ONE is connected to another MX-ONE or to other PBX-es which support equivalent network features by the way of an ISDN tie-line using QSIG, or via Digital Private Network Signaling System (DPNSS), or via H.323 or SIP tie-lines over an IP connection.

For information about which features that are available for the PSTN/ISDN and ISDN QSIG/DPNSS/ H.323/SIP connections, see the description for MIVOICE *MX-ONE FEATURE MATRIX*.

Note: Certain network features are only available when the MX-ONE is connected to another MiVoice Mitel MX-ONE, or certain other PBX models which support the protocol; these features may not work with a PBX from another vendor. The available features also differ a bit for the different signaling systems.

3.2.4 OPERATOR FEATURES

Operator features can be accessed from the attendant console/client.

For information on which operator features that can be initiated or terminated in NOW, see the description for *MIVOICE MX-ONE FEATURE MATRIX*.

3.3 SUPPORTED TERMINAL TYPES

The terminals described in this section are the ones that can be connected to MX-ONE. Most of these terminals are voice terminals.

3.3.1 IP PHONE AND IP CLIENT

Any SIP compatible IP terminal or SIP based soft phone/client, like MiCollab or BluStar may be connected to the MX-ONE. MX-ONE IP phones support the SIP protocol, for a list of the supported RFCs, see the *MIVOICE MX-ONE SYSTEM DESCRIPTION*.

Any H.323-compatible IP terminal (phone or PC client) can be connected to the MX-ONE. Basic functionality is available according to the H.323v2/v4 standard, including Inherent Free Seating. However, for full functionality, MX-ONE IP phones or Personal Assistant PC must be used, where a proprietary protocol is used on top of the H.323 protocol to get full functionality.

For more information on supported phones and clients, see the description for *IP EXTENSION*.

3.3.2 ANALOG PHONE

Any type of analog phone with DTMF signaling can be connected to the MX-ONE.

Rotary dialing phones are supported. The rotary dialing phones do not support DTMF signaling.

3.3.3 DIGITAL PHONE

An extension equipped with a digital phone can use the telephony features in an easier way than extensions equipped with analog phones. As the digital phones are equipped with pre-programmed keys for the most used features and programmable keys for other features (the most advanced also so called soft-keys and display), the features can be used without dialing procedures.

3.3.4 MOBILE PHONE

The mobile phone has access to features and functions comparable to an internal PBX extension, like callback, conference and so on.

To get the services and feature sets into the mobile phone, it is recommended that an agreement with a local mobile phone operator is in place.

3.3.5 CORDLESS PHONE

The cordless phone feature enables users to make and accept calls at any location in the coverage area of its base stations. The feature is fully compliant to the DECT

GAP/CAP standards, which ensures desk phone speech quality and full security from wiretapping.

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

3.3.6 SIP-DECT/IP-DECT PHONE

The Mitel SIP-DECT system and the alternative (Ascom) IP-DECT connect to the corporate IP network and act as gateways between the DECT portables and the IP network connected to the MX-ONE.

The main difference is that these base stations do not connect to the MX-ONE via a base station controller (e.g. ELU31) as is the case with integrated Cordless feature, but directly via the corporate IP network.

The DECT portables register via these Base stations as SIP extensions to MX-ONE. The SIP-DECT phone and IP-DECT phone features enable users to make and accept calls at any location in the coverage area of its base stations. The features are fully compliant to the DECT standards, which ensures desk phone speech quality and full security from wire-tapping. The SIP-DECT and IP-DECT handsets support a similar level of features as the integrated cordless feature, although the information on the display may vary a bit.

These solutions include full integration of optional alarm and messaging functions, which enable features such as messages to handset and messages or alarm from handset as well as Man-down notification.

3.3.7 EQUIPMENT USING CHANNEL ASSOCIATED SIGNALING EXTENSION INTERFACE

The Channel Associated Signaling (CAS) extension interface provides a digital connection to external equipment and offers them, through PCM lines, the functionality of analog extensions.

Besides the analog extension functionality, the CAS extension offers the ability to switch data from 64 kbit clear-channel data-interfaces connected to the external equipment. This is applicable, for example where a multiplexer (MUX) is used as a simple remote unit and permits connections from multiplexed data extensions to other MX-ONE data interfaces such as ISDN external lines and Digital Private Network Signaling System.

3.3.8 EQUIPMENT USING ISDN S0 TERMINAL INTERFACE

An ISDN S0 terminal has access or partly access to most of the MX-ONE specific extension features. The busy state related services such as Callback, Call Waiting, Diversion on Busy, Individual Call Pick-up, PBX group membership, Call Announcement, and Extending are however not supported.

An ISDN S0 terminal can be, for example, a telefax GP4, PC with ISDN board, PC with ISDN board and a phone, terminal adapter with a handset, video phone, Local Area Network, LAN, connection or an ISDN phone.

3.3.9 PC-BASED ATTENDANTS (NOW, ACS/MITEL INATTEND)

ACS/Mitel InAttend is an IP-based application for PBX operator for call handling and to search databases, and other functions relevant for PBX operators. The application is connected via SIP trunk and CSTA interfaces.

NOW is an IP-based application for PBX operators, used by the operators for call handling and to search the CMG database. There are also applications for handling of visitors and printing of internal phone books. The NOW provides directory services for the MX-ONE system. The operator may speak with either party individually or in conference. When the correct destination has been reached, the two parties are connected and the NOW is freed, ready to handle another call.

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

3.3.10 LEGACY OPI TERMINALS

The OPI-II, for example DGF 220 10, DBC 224, is an operators instrument for MX-ONE. The product is not available for new delivery, but can still be used with MX-ONE.

The OPI-II consists of the following two physical units:

- Visual Display Unit (VDU)
- A keyboard unit (OPI-K)

3.4 FEATURE LIST

3.4.1 ACCOUNT CODE

User Feature	**See also the description for *ACCOUNT CODE*.**
Definition	The primary purposes of the *account code* feature with verification are as follows: • To ensure extensions can charge calls to an account code (a project, department or client) instead of the extension’s own number • To prevent unauthorized use, the extension is forced to enter an account code before dialing an external number.
Settings	Account codes are set by the administrator.
Procedure and function	An extension dials a procedure and the account code prior to the destination number.

3.4.2 ACD/CTI BACKUP GROUP

System Feature	
Definition	The ability for an ACD or Computer Telephony Integration (CTI) group to use another ACD or CTI group as a backup group to prevent losses of ACD calls. This is done when the Server (Service Node), where the group resides, becomes unavailable due to that the Server is blocked, isolated or restarts. Any ACD/CTI group can be a backup group for another ACD/CTI group provided the two groups are in different Servers.
Settings	Backup groups are set by the administrator.
Procedure and function	When a Server (Service Node) is blocked, isolated or restarts and there is an ACD group in it that has a backup group, the following apply: • Delayed calls for the ACD group are deflected to the backup group • New incoming calls are diverted to the backup group

3.4.3 ADDITIONAL DIRECTORY NUMBER (ADN)

User Feature	
Definition	ADNs are extra directory numbers, beside the primary number, that can be programmed on a digital phone and are associated to a key. (See also Extra Directory Number, EDN).
Settings	Additional directory numbers are set by the administrator.
Procedure and function	It is possible to both receive and initiate calls on the ADN keys.

3.4.4 ADVICE OF CHARGE (AOC) / CHARGING DISPLAY INFO

User Feature	
Definition	Provides charging information (in currency) received from the public ISDN network and displays it on the charged extension, which must have an appropriate display. The information is conveyed through the private homogeneous ISDN/H.323 network. Only AOC requested at call setup is supported, and only the public ISDN services AOC-During and AOC-End are available. The charging display information provides the ability to display information about call cost on external calls on digital phones.
Settings	Class of Service (CoS) codes, and Charging Information are set by the administrator.
Procedure and function	An extension which has CoS for AOC, and makes a call using public ISDN, can obtain charging information and display it during the call and at the end of the call, or alternatively only at the end of the call.

3.4.5 ALARM EXTENSION

User Feature	
Definition	The ability to have an emergency center in the system. Alarm extensions can be assigned to analog extensions, ADN on digital extensions, cordless DECT extensions, CAS extensions and remote/mobile extensions.
Settings	Directory numbers for the alarm extensions are set by the administrator.
Procedure and function	If an alarm extension is busy when called, the call will be automatically conferenced with the alarm extension, except when: • The alarm extension or the party connected to the alarm extension, or the calling party, has a parked party • The alarm extension is not in speech state • The alarm extension is involved in an ordinary conference or another multi-party call Up to seven parties plus the alarm extension can simultaneously be connected to the same alarm conference. A periodic tone message called Conference Tone can be sent to all participants.

3.4.6 ALARM INDICATION

Operator Feature	
Definition	The ability for an operator to receive an alarm indication on the attendant console/client when a fault occurs in the exchange.
Settings	Allocation of fault types to specific alarm classes is done by the administrator.
Procedure and function	The attendant console/client.: Five different alarm classes can be displayed on the main window. A fault is indicated with a red colored indicator and a digit in the status bar. The operator can acknowledge the alarm and contact the service personnel. When the alarm is acknowledged, the indicator changes to yellow color.

3.4.7 ALTERNATIVE ROUTING

Network Feature	
Definition	The ability to reach external destinations through different routes.
Settings	Alternative routing and their pre-digits are set by the administrator.
Procedure and function	Every route can have seven alternative routes. The system uses sequential hunting on the ordinary route and the alternative routes, that is when the ordinary is fully occupied, the system starts hunting in the first alternative route and so on. The system can add and discriminate programmed pre-digits, that is, if a route to another PBX is fully occupied and the call has to be switched through the PSTN, the system adds the extra digits needed. A maximum of 20 pre-digits can be added to each alternative route, and the total code may not consist of more than 34 digits.

3.4.8 ANALOG EXTENSION

User Feature	
Definition	An Analog Extension is configured for a connection of a standard two-wire telephone with rotary dialing or DTMF signaling. The analog extensions supports that the phone number for the incoming call is presented on the called party's terminal display. This is done by using either of the two signaling methods to send out this type of information, DTMF tones or Frequency Shift Keying (FSK).
Settings	Analog extensions and their display configuration are initiated by the administrator.
Procedure and function	

3.4.9 AREA CODE PER EXTENSION/DOMAIN/GATEWAY/SERVER

System Feature	
Definition	An Area Code is a prefix to the directory number of an extension or other device. The area code may represent a geographical area in the public telephony network. In MX-ONE the area codes can be associated to four entities: the individual extension, the IP-domain, the media gateway, or the Server (LIM).
Settings	The area codes are set by the administrator. Area codes do not have to be set, i.e. can be omitted, but for certain features, like LCR and private networking they are required.
Procedure and function	The individual extension's home area code for a generic extension can be set in the extension initiation command. The area code for a Server (LIM), media gateway, or a domain where the Server or media gateway is located can be set with dedicated commands in the specific Server or media gateway. For IP/H.323 and SIP extensions, the area code where the extension is located can be associated to the domain representing that location, that is, the domain the extension belongs to. If all the area codes are set, and the originator of the call is a generic extension, the individual extension's home area code prevails over the media gateway area code and Server area code. The domain associated area code is not used, except in emergency calls and for the Call Admission Control function. If an H.323/SIP extension makes an emergency call, only the domain, media gateway or Server area codes are used. The domain associated code has higher priority than gateway and Server area code, and media gateway area code has higher priority than the Server area code. The resultant area code is referred to as the Own Area Code.

3.4.10 AUTHORIZATION CODE

User Feature	**See also the description for *AUTHORIZATION CODE FOR EXTENSION*.**
Definition	There are two types of authorization codes available to control or limit access to an extension: **Common Authorization Code** This code is affiliated to either all extensions within the system or a group of extensions. The code cannot be changed by the user. Two different functions are provided: • Locking/unlocking an extension. When locked, a lower common category code or common service profile is used. • Authorization code dialing. This enables the calling party to use other categories or service profiles than those with which the extension is programmed. **Individual Authorization Code (RAC)** This code is always affiliated to a directory number in the system. In addition to the functions available to common codes, individual codes also allow changing of the authorization code from the phone. This enables the user to change the authorization code when required. RAC can be used as pin-code for all IP extensions.
Settings	Authorization codes are set by the administrator. Individual authorization codes can be changed by the user.
Procedure and function	The user dials the allocated feature code.

3.4.11 AUTOMATIC CALL DISTRIBUTION FOR DIGITAL EXTENSIONS

System Feature	
Definition	Automatic Call Distribution (ACD) is an automated solution used to distribute a large quantity of incoming calls to a service controlled by the number dialed by the calling party. ACD can only be activated for digital extensions.
Settings	ACD groups are set by the administrator.
Procedure and function	Each service is connected to an ACD group consisting of one or more agents who handle these calls. Thereby it is possible to handle a large number of incoming calls without the need for PBX operators to route the calls. The agents are assigned as members of, and can answer calls from, one or more ACD groups. The selection of an agent can be based on selection priority and type of selection. When all the agents are busy, ACD places the incoming calls in queue and provides a call progress message.

3.4.12 AUTOMATIC NETWORK CALL DISTRIBUTION

System Feature	
Definition	The Automatic Network Call Distribution (ANCD) makes it possible to distribute incoming calls to different ACD groups based on the status of the ACD groups that are handling the required services. The involved ACD groups can be located in the same or different nodes. The distribution functionality can be used to distribute calls to the ACD group which provides the best answer capability for the moment, and it can also be used to redistribute calls from one ACD group to another at overflow situations. ANCD is a powerful complement to the basic ACD feature.
Settings	ANCD groups are set by the administrator.
Procedure and function	

3.4.13 BASIC CALLS

Operator Feature **Network Feature** **User Feature**	
Definition	Basic calls consists of functions for external and internal calls. Users with any type of phone can make and receive calls. Certain phone types can make and receive calls on different lines. For example, on some digital or IP phones a second call can be accepted if the function key Free on Second Access is activated. Features: • Calling another extension or group number. See also Internal Basic Calls. • Calling on another line • Call to Individual External line • Receive Call (Normal Case) • Receive Second Call on Same Line • Receive Second Call on Another line. See also Free on Second Access.
Settings	-
Procedure and function	Internal Call: The extension lifts the handset and dials the required number on receipt of the dial tone. If the extension fails to dial within a certain time (8 s) or the pause between two digits exceeds a certain time (8 s), a disconnection signal is sent.

3.4.14 BOSS-SECRETARY

User Feature	**See also the description for *BOSS-SECRETARY*.**
Definition	This feature provides secretarial screening by one or more secretaries for one or several bosses. When the feature is active, calls to the boss are directly forwarded by using a specific Individual Repeated Distribution (IRD) list to one or more secretaries. The feature is often complemented with secretarial monitoring where assigned monitoring keys on IP phones and digital phones shows the status (free, ringing, parked or busy) of the monitored phones, indicated by a Light Emitting Diode (LED). The manager or the secretary can also use the programmed monitoring key to call the monitored party. See also MNS and MDN features in this document. Secretaries can by-pass the feature if their directory number is included in the boss Individual Repeated Distribution (IRD) list that is used for this purpose. When a secretary is calling a boss, and her number is not part of the boss Individual Repeated Distribution (IRD) list the call is treated as normal call (that is, the call is deflected to that boss' secretary).
Settings	Initiated by the administrator.
Procedure and function	While the feature is active, the calls to the boss phone will be deflected to the secretary's phone, and the corresponding PEN key LED (for DTS and IP extensions) will indicate that the service is on for both the boss and secretary phones.

3.4.15 CALL ADMISSION CONTROL

System Feature	**See also the operational directions for *CALL ADMISSION CONTROL*.**
Definition	The purpose of the *Call Admission Control (CAC)* feature is to manage the available bandwidth used for voice calls through low bandwidth links. This type of links is usually used to connect branch offices to the main office and due to the restricted bandwidth that these links normally have, a mechanism is needed to manage them properly in order to maintain a certain quality of service in the calls when media go through these kind of links.
Settings	The domains and the codec priorities can be initiated or modified by the administrator.
Procedure and function	Call Admission Control only applies to gateway calls and inter-domain non-gateway calls. CAC is valid for all calls requiring RTP resources, that is, IP extensions (both SIP and H.323), IP trunks, IP operators, and calls using the IP group switch. There is no end user action involved. The CAC data is defined per network domain. Bandwidth is considered free for calls within a domain. If one of the RTP-addresses belongs to a domain with bandwidth limitations, a CAC reservation will be done. If the bandwidth is inadequate the call will be rejected. For non-gateway calls, the codec negotiation is done directly between the network endpoints. The result of the negotiation will be sent for CAC evaluation and the bandwidth will be adjusted accordingly.

3.4.16 CALL ANNOUNCING

Operator Feature	
Definition	The ability for an Operator to announce incoming calls to a busy extension.
Settings	-
Procedure and function	The operator uses a specific key for manual ringing if the extension is free. A specific supervision link can be used if the extension is busy. The operator is hereby recalled as soon as the wanted extension becomes free and the operator uses the key for manual ringing. The extension must be in the operator's exchange.

3.4.17 CALL CENTER GROUP (CTI GROUP)

System Feature	
Definition	The Call Center Group (a.k.a. Contact Center group or CTI group) is a class of service for the Automatic Call Distribution (ACD) feature. Calls to a CTI-group are queued by the system and information about queued, distributed and forwarded calls are sent through the CSTA interface, to an external application which controls the routing of the calls, using call deflection or other CSTA services.
Settings	CTI groups and their configuration are initiated by the administrator.
Procedure and function	-

3.4.18 CALLBACK, BUSY EXTENSION/CALL COMPLETION

User Feature	
Definition	The ability to initiate a supervision on a busy extension, and be automatically rung when the dialed extension becomes free.
Settings	For normal use, there is no CoS for Callback, but the special CoS for ‘automatic callback request at calls from an external device’ is set by the administrator.
Procedure and function	The calling party dials a suffix digit when the busy tone is received. A confirmation tone is sent to acknowledge the supervision. Both parties are supervised and, whenever concurrently free, a connection is established. The calling party is rung with a special Callback ringing signal. If the calling party cannot be alerted, or does not answer within a predefined time, the supervision is either (if possible) re-initiated (as a reverse Callback) or canceled. Manual cancellation of a Callback order is done via a procedure from the phone.

3.4.19 CALLBACK, BUSY OUTGOING ROUTES/TRUNKS

User Feature	
Definition	The ability to initiate supervision on a busy route (or individual trunk line) and be automatically rung when an external line becomes free.
Settings	For normal use, there is no CoS for Callback, but the special CoS for 'automatic callback request at calls from an external device' is set by the administrator.
Procedure and function	The extension dials a suffix digit when the busy tone is received. The dial tone is sent to the extension which now dials the route number plus all or part of the external number completed by #, and replaces the handset. The extension is rung, if free, when an external line becomes free and all the digits dialed except the last. The last digit is sent when the extension answers the Callback. If the extension does not answer within a predefined time, the supervision is canceled. Cancellation of a Callback order is done with a procedure from the phone.

3.4.20 CALLBACK, NO REPLY/CALL COMPLETION

User Feature	
Definition	The ability to establish supervision of an extension that does not answer a call.
Settings	For normal use, there is no CoS for Callback, but the special CoS for 'automatic callback request at calls from an external device' is set by the administrator.
Procedure and function	Supervision is established when the calling extension dials a suffix digit when the ringing tone is received. Both parties are supervised. The calling party is rung with a special Callback ringing signal as soon as the called party replaces the handset after having used the phone. If the calling party cannot be alerted, or does not answer within a predefined time, the supervision is either (if possible) re-initiated as a reverse Callback, or canceled. Manual cancellation of Callback is done with a procedure from the phone.

3.4.21 CALLBACK, NOT AVAILABLE/CALL COMPLETION

User Feature	
Definition	The ability of an extension in one system to initiate supervision of an unavailable extension, and to be rung automatically when the called extension becomes available and free.
Settings	For normal use, there is no CoS for Callback, but the special CoS for 'automatic callback request at calls from an external device' is set by the administrator.

User Feature	
Procedure and function	An extension which has called an unavailable extension in a terminating exchange can, while receiving busy tone, initiate Callback/Call Completion by means of a suffix digit. When the supervised party becomes available, the calling party is called back. When the calling party answers, ringing starts at the called party. If the calling party cannot be alerted, or does not answer within a predefined time, the supervision is either (if possible) re-initiated as a reverse Callback, or canceled.

3.4.22 CALLBACK, FAULTMAN'S RING BACK

User Feature	
Definition	This type of callback allows a test of the phone equipment. Faultman's Ring Back is only possible internally within a PBX. The initiation is done by dialing the own extension number (initiating party is the same as called party) and when the busy tone is received, dial the procedure for Callback. The extension CoS for internal callback is checked. When the extension goes on hook the extension is recalled immediately. At answer, a message is sent to the initiating party indicating that this is a Faultman's callback call.
Settings	-
Procedure and function	Requested as Callback on busy when busy tone is received. The Faultman's ring back (recall) will be executed immediately on onhook after the request.

3.4.23 CALL INFORMATION LOGGING (CIL) AND QOS LOGGING

System Feature	See also the description for *CALL INFORMATION LOGGING, QUALITY OF SERVICE LOGGING.*
Definition	The MX-ONE provides data for all types of calls. Data records for the calls are generated in the MX-ONE Service Node (SN) and, after the end of the call, the required data is stored in any of the available output formats and sent to any of the possible output ports. Abandoned incoming trunk calls due to no answer are also registered. The following data are examples of what is logged: • Time at start of call • Time at end of call • Call duration • Date • Charging information, either as a cost in a currency amount or as number of meter pulses, if a pulse receiver exists. • Identity of calling party (extension or operator directory number), incoming external line number or, if an authorization code has been entered, the extension number affiliated to the code. • Dialed external line route number • Dialed number (maximum 20 digits) • Quality of Service data for IP (H.323) traffic.
Settings	Call criteria (that is, which types of calls should be logged) are set by the administrator. The QoS logging feature can be turned on/off.
Procedure and function	-

3.4.24 CALLING/CONNECTED LINE IDENTITY

User Feature **Network Feature**	
Definition	Automatic presentation of the calling or connected party’s number to the called party’s exchange.
Settings	-
Procedure and function	During call setup, the calling or connected party’s number is transmitted automatically to the called party and to the called party’s exchange. The connected line identity feature is used after a transfer or when a conference has returned to a two-party conversation, the remaining parties are updated with the correct number information. A general number presentation restriction is supported, and is activated when the extension is initiated. The restriction is then active for all calls.

3.4.25 CALLING LINE IDENTITY RESTRICTION

User Feature **Network Feature**	
Definition	The ability of a calling party to restrict presentation of the calling party’s number to the called party and to the called party’s exchange. Different settings can be valid for internal and external calls. There is also a presentation restriction override function.
Settings	Presentation restriction is set by the administrator. Override permission is set by the administrator.
Procedure and function	During call setup, the calling or connected party’s number presentation restriction is transmitted automatically to the called party and to the called party’s exchange. The restriction can be overridden by users with override permission. PBX operator will always override any presentation restriction.

3.4.26 CALLING LINE IDENTITY RESTRICTION - PER CALL

User Feature	
Definition	The ability of a calling party to restrict presentation of the calling party’s number to the called party’s exchange on a per call basis.
Settings	-
Procedure and function	The caller may initiate number presentation restriction for an individual call using pre-dial procedure.

3.4.27 CALLING/CONNECTED/ROUTE NAME IDENTITY

User Feature	**See also the description for *NAME IDENTITY.***
Definition	The *Name Identity* feature is used to associate an easily recognizable name with various individual items that are normally identified in the system by a number. The feature is also useful for incoming and outgoing external calls where the external number exceeds the length of the field reserved for it in the display. The *Calling/connected name identity* feature allows the assignment of a name to the following entities/devices: • Analog extensions (primary and secondary) • Hunt Groups • IP terminals • Digital extensions • Remote/mobile extensions The feature also allows the name associated with an individual to be conveyed through the system, together with its number, so that both the number and name are displayed together. The name identity of the parties involved in a call is transferred. For external calls this is by way of an ISDN private or public network. Name presentation restriction is supported, but if the *Calling/Connected Line Identity Restriction* is activated, it takes precedence over the name presentation restriction. The *Name on Route* feature is used to associate an easily recognizable name to the route. The feature is also useful for incoming and outgoing external calls where the calling/connected/called name identity is not received.
Settings	The names are set by the administrator.
Procedure and function	-

3.4.28 CALL METERING/CHARGING

System Feature	
Definition	The ability to detect, store and read stored metering pulses from a public exchange.
Settings	The metering routes and lines are set by the administrator.
Procedure and function	External lines can receive metering information if provided by the public exchange. The cost/metering pulses are stored per extension or per authorization code.

3.4.29 CALL METERING/MAXIMUM CALL COST

System Feature	
Definition	The ability to set a maximum limit on the received metering pulses from a public exchange.
Settings	The metering limit is set by the administrator.
Procedure and function	External lines can receive metering information if provided by the public exchange. The cost/metering pulses are stored per extension or per authorization code.

3.4.30 CALL PICK-UP, COMMON BELL GROUP

User Feature	
Definition	The ability for any extension within a defined group to pick-up calls to the group. The calls are indicated on a common signal device (for example, a bell).
Settings	Common bell groups are set by the administrator.
Procedure and function	Incoming calls are answered by dialing a service code. Incoming calls can be queued and as long as waiting calls exist they are signaled. Calls to the group can be diverted either directly or by the Follow-me procedure.

3.4.31 CALL PICK-UP, INDIVIDUAL

User Feature	
Definition	The ability to answer a call to any other extension from any phone.
Settings	-
Procedure and function	A call to an extension can be answered from any phone by dialing the extension number followed by a suffix digit when the busy tone is received.

3.4.32 CALL PICK-UP, GROUP

User Feature	**See also the description for *EXTENSION GROUPS*.**
Definition	The ability of a member of a defined group to pick up a call to another member in the group.
Settings	The initiation of groups and alternative answering groups (if any) are done by the administrator. The optional display function for other members than the one ringing can be configured for SIP extensions by the administrator.
Procedure and function	Calls to one extension within a group can be answered by any other group member by dialing a service code. Each group can have four alternative answering groups and, if no calls to its own group exist, calls to the alternative groups are answered with the same procedure. It is not possible to answer Callback to another party member, nor is answer permitted if both calling and answering parties have a parked party. There is an optional display function for other group members that are SIP extensions, which can show caller number and name and called party of the oldest ringing group call for the potential picking party.

3.4.33 CALL SPLITTING

Operator Feature	
Definition	The ability to initiate a three-way speech connection where the operator can listen to both connected parties, then put one on hold and talk to the other.
Settings	-
Procedure and function	Three-way speech can be initiated with a specific key, when the operator has answered calls on both sides. Call splitting is done by pressing one of the speech direction keys while in a three-way speech situation.

3.4.34 CALL TO INDIVIDUAL EXTERNAL LINE

Operator Feature User Feature	
Definition	The ability for an operator to select a specific line for outgoing calls.
Settings	-
Procedure and function	The user can by a procedure select on individual external line. Note that non-generic extensions are allowed to dial the procedure, but would have to know the trunk identity in advance, since it is not displayed.

3.4.35 CALL WAITING/CALL OFFER

Operator Feature User Feature Network feature	
Definition	The ability of an extension or the operator to send an audible indication that an internal or external call is waiting. Call offer from an operator is only applicable if ISDN QSIG, DPNSS, SIP or H.323 signaling is used in the private network.
Settings	The ability to send and receive a Call Waiting tone is given individually by the CoS and set by the administrator.
Procedure and function	A calling extension initiates Call Waiting Indication by dialing a suffix digit. Call Waiting Indication is automatically sent on calls routed via the operator or on Direct In-Dialing lines, if this is programmed. The ring tone is sent for 30 seconds to the calling party and the extension can during this time answer the waiting call by terminating, parking or transferring of the ongoing call.

3.4.36 CAMP ON BUSY

Operator Feature	
Definition	The ability of the operator to camp on a call to a busy extension.
Settings	-
Procedure and function	The operator extends the call in the normal way and the waiting call is indicated to the called party. The call is connected automatically to the extension when the call in progress has finished. The call is routed back to the operator if not answered within 60 seconds. The operator may camp several calls on the same busy extension. On a call to a busy extension in the network, a check is made to verify if the incoming external line category permits network services and if the external originating party is an operator. If so, the call is camped on in the terminating exchange. When an operator with a connected party calls a busy extension in the terminating exchange, a message is provided stating whether extending of the connected party is possible.

3.4.37 CAS EXTENSION INTERFACE

User Feature	
Definition	The CAS extension interface provides a digital connection to external equipment and offers them, through PCM-links, the functionality of analog extensions. The functionality of the analog extension is only limited depending on the connected external equipment, for example, if the external equipment does not support hook-flash or message waiting. The CAS extension supports old Voice Mail systems in upgrading scenarios.
Settings	CAS extensions are initiated by the administrator.
Procedure and function	-

3.4.38 CENTRALIZED ANSWERING POSITION (C-AP OR C-OP)

Operator Feature	**See also operational directions for *CENTRALIZED ANSWERING POSITION*.**
Definition	Centralized Answering Position is a collection of functions for having centralized answering positions in one exchange that can serve several other exchanges. Typical included functions are: • Status notification (day/night operation mode) regarding centralized answering position from one exchange to the other exchanges (only for operators). • Night service diversion to the centralized answering position. • Rerouting of incoming trunk traffic to the centralized answering position. Various supplementary services across the network, like Intrusion or Supervision (Call Back) are also included in the centralized operator functionality. Operators, extensions and extension groups can be defined as central answering positions. Note that if the centralized answering position is an extension or an extension group, then status notification is not supported which limits the number of applicable functions.
Settings	The function is set up by the administrator.
Procedure and function	-

3.4.39 CHOICE OF LANGUAGE

User Feature	**See also the description for *CHOICE OF LANGUAGE*.**
Definition	The *choice of language* feature allows each user of a digital phone or an MX-ONE IP phone to select the language in which the text messages will appear. On a digital phone, the user can select one of ten available languages by dialing a procedure from the phone. On an IP phone, the user selects the language by either selecting a language in the phone menus or by dialing a procedure from the phone. The system administrator can specify the language for the IP terminal from the ten available languages or set it as the exchange language.
Settings	The language of text strings shown on the digital or IP phone's display can be selected on a per-user basis.
Procedure and function	Any user can change the language for the extension by dialing a procedure from that particular terminal.

3.4.40 CLASS OF SERVICE (COS) OR COMMON SERVICE PROFILE (CSP)

System Feature	**CoS for analog and digital extensions, CSP for generic extensions.**
Definition	The ability to give every extension an individual CoS corresponding to allowed or denied types of calls and/or facilities.
Settings	The CoS and CSP is set by the administrator.

System Feature	CoS for analog and digital extensions, CSP for generic extensions.
Procedure and function	The CoS consists of common classes and individual classes. The CSP consists of only common classes. Common classes are assigned as system parameters and affect all extensions in the system. Individual classes are given per extension and open up or close the possibility of using facilities, for example, Intrusion.

3.4.41 COMMON PUBLIC DIRECTORY NUMBER

System Feature	
Definition	The ability for an ISDN connected public exchange to receive the identity of the connected A-party in the calling party number. The identity is used for charging in the public exchange. To avoid charging the wrong subscriber the identity of A-party must be included in the number series the system has been assigned in the public exchange. The common public directory number is used when charging is to be carried out on a common number because the extension number is not allowed to be transmitted to the public exchange or the extension number is not available.
Settings	The common public directory numbers are set by the administrator. Can be set per system or per Server (LIM).
Procedure and function	-

3.4.42 COMPUTER SUPPORTED TELECOMMUNICATIONS APPLICATIONS (CSTA), PHASE 1

System Feature	See also the description for *COMPUTER SUPPORTED TELECOMMUNICATIONS APPLICATIONS (CSTA WITH APPLINK), CS.*
Definition	CSTA Phase 1 (Application Link) is an application protocol that enables a computer domain to communicate with a telephony domain. The ASN.1 protocol is supported. CSTA enables third party call control, and is used for example for Call Center applications. **Note:** No further development of the CSTA Phase 1 is intended.
Settings	The CSTA interface is set up by the administrator.
Procedure and function	When a call arrives, a message is sent from the MX-ONE to the connected computer application to inform it of the event. The message contains information such as which extension received the call, who is calling and which number was dialed. Only device monitoring is supported, and event reporting is done per device. The application can send service requests to the MX-ONE, and get telephony services executed. Outbound call center calls can be supported.

3.4.43 COMPUTER SUPPORTED TELECOMMUNICATIONS APPLICATIONS (CSTA), PHASE 3

System Feature	**See also the description for *CSTA PHASE III***
Definition	CSTA Phase III is an application protocol, which is an enhancement to CSTA phase 1, that enables a computer domain to communicate with a telephony domain. Both the XML protocol and XML via WebServices or via SIP (TR87) is supported. CSTA enables third party call control, and is used fore example by Call Center applications, such as *MiContact Center Enterprise*. **Note:** Support for CSTA3 with ECMA 285 ASN.1 has been withdrawn.
Settings	The CSTA interface and its configuration is set up by the administrator.
Procedure and function	When a call arrives, event messages are sent from the MX-ONE to the connected computer where a CSTA application is executing, to inform it of the event(s). An event message contains information such as which extension received the call, who is calling and which number was dialed, and which service is invoked. Application Session Services (authentication of the application) are supported. Both monitoring of device (most types of devices) and monitoring of call is supported. The application can send service requests to the MX-ONE, and get telephony services executed. Inbound and Outbound call center calls can be supported, but also any basic call.

3.4.44 CONFERENCE

User Feature	**See also the description for *CONFERENCE*.**
Definition	The ability to establish conference calls with 3 - 8 parties.
Settings	The number of permitted participants is set by the administrator. The conference warning tone can be turned on/off by the administrator.
Procedure and function	This type of conference call can either be set up by an extension or by the operator, provided the category of participants allows Conference calls. The participants can be an internal party, a private network party or an external party in any combination within the limitations stated above. An optional warning tone is issued to all participants at regular intervals during the call. Only the conference leader in an extension initiated conference can use the *inquiry* and *refer back* features, and receive indications that another call is waiting during the conference. The conference leader can, by using the *inquiry* and *intrusion* features on one of the participants, temporarily place both parties outside the conference. One or both can return to the conference depending on the procedure used by the conference leader. Display information indicating the number of conference participants, and the role (conference leader/member) is available for SIP terminals (if they support it).

3.4.45 CORDLESS EXTENSION (DECT)

User Feature	
Definition	The cordless extension is an extension type that enables users to make and accept calls at any location in the coverage area of its base stations by using radio signals. The cordless extension is fully compliant to the Digital Enhanced Cordless Telecommunication (DECT) GAP/CAP standards, which ensures desk phone speech quality and full security from wiretapping. In addition to the integrated cordless extension, two other versions of cordless extension solutions, called SIP-DECT and IP-DECT, are also available. The base stations are in those solutions connected to the system as SIP extensions.
Settings	Cordless extensions are initiated by the administrator. The SIP-DECT and IP-DECT are also initiated by the administrator, see IP extension (SIP).
Procedure and function	-

3.4.46 CORPORATE LOG-ON

User Feature	
Definition	The corporate log-on feature (also called network roaming) is available when configured for Mitel 6900/6800/6700 SIP terminals and H.323 terminals (Mitel 7444ip only). It means the extension is registered in its home-PBX, even when it is a guest extension in another PBX in the private network.
Settings	Configuration files for the terminal types must be appropriately set to support corporate log-on.
Procedure and function	The SIP or H.323 extension user uses the ordinary log-on procedure to register, and log-off procedure to log off, also when located in another node than the home PBX, in the private network.

3.4.47 CUSTOMER GROUP

System Feature	See also operational directions for *CUSTOMER GROUP*.
Definition	An MX-ONE can consist of a number of customers (tenants) who can be separated with regards to telephony. The customers are arranged in one customer group. This feature is also known as a Multi-Tenant Group.
Settings	Customers are initiated by the administrator.
Procedure and function	Each customer virtually has their own system. Customers (tenants) within the customer group can have their own resources, such as routes and operator groups, but also have features for utilizing common system resources within the customer group. Each customer can have its own number plan.

3.4.48 CUSTOMER IDENTITY STORAGE (CID)

User Feature **Network Feature**	
Definition	This feature allows a Customer Identity to be associated with an external caller (customer).
Settings	-
Procedure and function	The identity may be received automatically, or manually entered by the calling party using the DTMF keypad. If the call is transferred, for instance via a voice server, the Customer Identity is transferred with the call.

3.4.49 DATA PRIVACY

User Feature **Operator Feature**	
Definition	The ability for an extension to be protected against features such as *intrusion* and *call waiting* for the duration of the call.
Settings	-
Procedure and function	By dialing a procedure prior to the call, the Data Privacy is activated. An attendant can activate/deactivate the feature for an extension. **Note:** The Data Privacy feature is only available for non-generic extensions. Generic extension can use the Do Not Disturb feature to get a similar function.

3.4.50 DAY/NIGHT SWITCHING

Operator Feature	
Definition	The condition for the *Day/Night Service* feature can be specified to be either depending on the time of day or depending on the presence or absence of an operator. The setting of the Day/Night Service condition is necessary, for example, for the Day/Night Service CoS function.
Settings	The conditions are set by the administrator.
Procedure and function	The feature is activated by dialing a prefix procedure from the operator.

3.4.51 DIAL BY FUNCTION KEY

User Feature	
Definition	The ability to store an internal or external number on a key on a digital phone or an IP phone.
Settings	Set by the user.
Procedure and function	When pressing a specific key the stored number is automatically called.

3.4.52 DIAL BY NAME

User Feature	**See also the description for DIAL BY NAME.**
Definition	The dial by name feature allows a user to initiate a call by entering the other party's name, or just the beginning of it, via a standard keypad with letter designations and soft-keys. The feature provides a directory of names with their associated phone numbers. Numbers in the directory can be internal or external. The dial by name feature is only available for the digital phones with a standard keypad with letter designations and soft-keys. The same feature is normally available in the form of contact lists in intelligent telephones
Settings	The names are set by the administrator.
Procedure and function	The digital phone user interface allows searching in the name list by use of the keypad and soft-keys. Two of the soft-keys are used to scroll up or down in the list and one soft-key is used to execute the call when the requested name is found.

3.4.53 DIAL DURING A CONNECTED CALL (SUFFIX DIALING)

User Feature	
Definition	The ability of an extension in speech state to dial suffix digits/characters, for example, to control an external equipment.
Settings	How the suffix dialing shall be treated is set by the administrator.
Procedure and function	The feature is automatically available to analog extensions and IP extensions in speech mode, but PBX operators, digital and cordless extensions must request DTMF mode manually, except when the connected party is a voice mail machine.

3.4.54 DIALED NUMBER INFORMATION SERVICE (DNIS)

System Feature	**See the description for *DNIS*.**
Definition	The ability for the CTI groups to identify different customers based on the number the customer is dialing. Primarily used by Call Center applications for routing purposes.
Settings	The DNIS numbers are initiated by the administrator.
Procedure and function	When the CTI group call is presented, the DNIS number and name are displayed together with the calling party number and name. Each DNIS number is stored together with its affiliated name and CTI-group number. An incoming DNIS call fetches its stored name and the call is sent to its specified service group number. When the call is presented to the receiver, the DNIS number and name are presented.

3.4.55 DIGITAL EXTENSION

User Feature	
Definition	A digital phone equipped with preprogrammed keys for the most used features and programmable keys for other features. These features can be used without dialing procedures. It is possible, by command, to prevent automatic answer from the phone.
Settings	Digital extensions are initiated by the administrator.
Procedure and function	-

3.4.56 DIRECT-IN-DIALING

System Feature	
Definition	The incoming public external line calls can be routed directly to extensions with Direct-In-Dialing.
Settings	Type of signaling indication of origin and rerouting in certain traffic cases are set by the administrator.
Procedure and function	The extension number is transmitted from the public exchange. The digits are analyzed to find out whether the extension has a CoS that allows *Direct In-Dialing.* **NOTE:** The call can be rerouted (for example, to an operator who gets information about the dialed number and reason for rerouting) if congestion occurs, or if the extension is not answered, busy, vacant, blocked for *Direct In-Dialing* or in line-lockout state.

3.4.57 DIRECT-IN LINES

System Feature	
Definition	The ability to program incoming calls on manual external lines for direct connection to predefined extensions, Hunt groups or operator.
Settings	Set by the administrator.
Procedure and function	Every manual external line in the system is given one day address and one night address. The address can be an individual call number or a group number. A direct-in line that is not answered within 30 seconds is rerouted to the operator.

3.4.58 DIRECT INWARD SYSTEM ACCESS (DISA)

System Feature	
Definition	Direct Inward System Access is a facility allowing external users (voice calls) to call in to a PBX and get access to a limited set of the PBX's features. The primary application is the ability for users to charge expensive long distance calls to the PBX. A DISA call can be established by the use of Direct In-dialing external lines or manual external lines. These lines have to be connected to a DTMF code receiver to provide end-to-end signaling.
Settings	The DISA is configured by the administrator.
Procedure and function	A DISA number is a unique number within the numbering plan for that specific PBX. DISA is accessed by dialing this unique number followed by the feature code, the authorization code and the desired number. There is only one call access to DISA. After a call has been terminated, the user must clear down before the next call setup. All costs incurred are charged to the Call Information Logging code tied to the common authorization code dialed. The authorization code can be omitted.

3.4.59 DIVERSION, BUSY

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability for extension types of terminals with Individual Diversion to have calls diverted when encountering a busy state.
Settings	The CoS is set by the administrator. Diversion can be activated automatically at the initiation of an Individual Diversion position.
Procedure and function	Diversion is activated and canceled by the extension or the operator. Calls to a busy extension with Individual Diversion are diverted to the answering position. Note that an alternative feature can be *Personal Number* (with busy option).

3.4.60 DIVERSION, BYPASS

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability to bypass the diversion state.
Settings	-
Procedure and function	The programmed answering position can always reach the diverted extension by dialing the diverted number. An extension that has a CoS allowing Intrusion is also allowed to use Diversion Bypass. The function works internally and within a private network using ISDN, DPNSS, SIP or H.323 signaling.

3.4.61 DIVERSION, COMMON

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability of an extension to have calls diverted to a common answering position.
Settings	The CoS and the allocation of an answering position are set by the administrator.
Procedure and function	Diversion is activated and canceled by any authorized user. A diverted extension that initiates a call receives a special dial tone and visual indication (LED or display, if available) to indicate that the diversion state prevails.

3.4.62 DIVERSION, DIRECT

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability of an extension to temporarily move incoming calls to another (extension) position or operator.
Settings	The CoS and the diversion destination is set by the administrator. Diversion can be activated automatically at the initiation of an Individual Diversion position.
Procedure and function	Diversion is activated and canceled by any authorized user. The diversion destination is set from the command line interface. Calls to a free or busy extension with Direct Diversion are diverted to the answering position.

3.4.63 DIVERSION, INDIVIDUAL

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability of an extension to have calls diverted to an individual answering position. The answer position selection can be dependent on call origin (public, private, internal) and the call is then diverted to either the individual position or to the common position specified for that origin.

User Feature	See also the description for *CALL DIVERSION*.
Settings	The CoS and the allocation of answering positions are set by the administrator.
Procedure and function	Diversion is activated and cancelled by any authorized user. A diverted extension that initiates a call receives a special dial tone to indicate that the diversion state prevails.

3.4.64 DIVERSION, NO REPLY

User Feature	See also the description for *CALL DIVERSION*.
Definition	The ability of an extension with Individual Diversion to have calls diverted on No Reply.
Settings	The CoS is set by the administrator.
Procedure and function	A call to an extension with Individual Diversion that is not answered within 14 seconds is diverted to the answering position. Subsequent calls are diverted after 8 seconds provided the extension has not initiated a call in the meantime. Note that *Personal Number* can be an alternative to this feature.

3.4.65 DIVERSION, ON ORIGIN

System Feature	See also the description for *CALL DIVERSION*.
Definition	The ability of extension users to divert their calls to different answering positions depending on the origin of the call, that is, if it is an internal, external, or private network call.
Settings	The feature uses common or individual diversion numbers and an extension CoS, which are set by the administrator.
Procedure and function	Calls to an extension which has Diversion On Origin can be diverted to three different numbers depending on the origin type. Diversion can also be avoided for example for one origin. The normal diversion procedures are valid. Note that *Personal number* can be used as an alternative to this feature.

3.4.66 DIVERSION, VERIFICATION OF DIVERSION

User Feature	See also the description for *CALL DIVERSION*.
Definition	The presentation to users that the extension has an active diversion.
Settings	-

User Feature	See also the description for *CALL DIVERSION*.
Procedure and function	For a digital phone with display and active direct diversion, the number of the divertee position is displayed and the Diversion lamp indicates that the triple access line is diverted. For IP and digital phones with display and active Follow-me, Follow-me and the divertee position is displayed. For external follow-me, only a text is displayed. For IP and digital phones with display and active message diversion, absence reason and time back is displayed. For all telephones, a special dial tone indicates that diversion is activated when going off hook.

3.4.67 DO NOT DISTURB, BYPASS

Operator Feature	
Definition	The ability of an operator to bypass the do not disturb state.
Settings	-
Procedure and function	Bypass of DND can be done with the same procedure as Diversion, Bypass.

3.4.68 DO NOT DISTURB, GROUP

User Feature	See also the description for *EXTENSION GROUPS*.
Definition	The *group do not disturb* feature allows a master extension or an extension with GroupDoNotDisturb programming category set to mark a group of extensions as Group Do Not Disturb, that is, calls to extensions in the group are not signaled on the phone.
Settings	Groups and group members are initiated by the administrator. There is also a class of service controlling if activation/deactivation of GDND shall be allowed from an extension.
Procedure and function	If any extension in the group has any diversion activated or an individual divertee position, the call is diverted. If the extension has no diversion and the incoming call is a Direct In-Dialing call which has a CoS permitting rerouting, it is rerouted to an operator. 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.Bypass of Group Do Not Disturb can be done with the diversion bypass procedure from a master extension or an extension with Group Do Not Disturb programming category set.

3.4.69 DO NOT DISTURB, INDIVIDUAL

User Feature	
Definition	The ability of an authorized extension to prevent calls from being signaled at the phone set.
Settings	The Individual DND service as such has no settings, but if Personal Number is used, there is a setting for a DND re-direction destination.
Procedure and function	An extension can invoke the feature by dialing a procedure. No further calls to the extension are permitted, and Direct In-Dialing calls are rerouted, for example, to the operator.

3.4.70 DPNSS NETWORKING

Network Feature	**See also the description for *NETWORKING*.**
Definition	DPNSS is an open standard signaling system for integration of PBXes in a network. DPNSS networking allows communication between the MX-ONE and other telephony systems using DPNSS signaling.
Settings	The DPNSS networking is set up by the administrator.
Procedure and function	DPNSS networking allows that a telephony system can be connected through leased lines to remote PBXes in multi site organizations or to PBXes of other organizations. If a DPNSS network is connected to a QSIG network, only basic calls are allowed at the gateways.

3.4.71 DSS1 NETWORK SIDE

System Feature	**See also the description for *DSS1 NETWORK SIDE*.**
Definition	The MX-ONE can act as a network side of an ISDN-T connection. This is mainly used for branch office applications, where the branch office acts as the user side.
Settings	The DSS1 network side is set up by the administrator.
Procedure and function	-

3.4.72 EMERGENCY CALLS, SOS CALLS

User Feature	**See also operational directions for *EMERGENCY CALLS, SOS CALLS*.**
Definition	The IP phones can call an emergency number when they are logged on. Some terminal types can also call the emergency number when logged off by setting up the emergency call to a specific H.323 or SIP access of MX-ONE. The call is given the highest priority. The MX-ONE sets up the call to an emergency center through an external public destination defined as emergency destination. The A-number of the IP phone, or a Location identity, can be stated by the administrator and is preferably defined according to geographical location. The call will be routed to an emergency center located in the same area as the calling user, to receive help as fast as possible. This feature should not be confused with the facility to call an international emergency call number (for example, 112 or 911) that all extensions are able to call for assistance. For Mitel 6900/6800/6700 SIP phones also the Emergency Location Identification Number (ELIN) and a Location Identity is supported. These numbers can be used as a call ID and caller ID or separate Location Identity (SIP PANI header) in emergency calls, allowing enhanced location information. A similar function is supported for H.323 phones.
Settings	Emergency numbers and area codes are set by the administrator. The location Identities are also set by administrator, per IP domain. Both the IP phones and the Service Node need configuration for the Emergency calls.
Procedure and function	Once a user with an IP phone has made a emergency call, the emergency center is able to dial back to the extension(s) that initiated the emergency call(s). The location data can be conveyed in a few different ways, as Caller Name, Caller User number or as a separate Location Identity (SIP PANI Header). An emergency call can optionally generate an SNMP trap and an alarm which can be used by external applications (e.g. Mitel Mass Notification or Mitel Performance Analytics) for further measures. See the SNMP feature.

3.4.73 EMERGENCY CALLS TO OPERATOR

User Feature	
Definition	A special type of incoming call to the operator which is given the highest priority.
Settings	Emergency operator numbers are set by the administrator.
Procedure and function	Extensions and tie-lines can initiate calls to operator with the highest priority by dialing a predefined emergency operator number. These calls will bypass the operator queues of ordinary calls.

3.4.74 EMERGENCY NOTIFICATION

User Feature	
Definition	Broadcast an emergency message to a group of extensions. The supported extension types are analogue, digital and IP terminals.
Settings	Emergency Notification numbers are set by administrator.
Procedure and function	While the feature is active, the calls to the emergency notification number will be processed according to dedicated personal number list positions which are associated to group hunting group without members. The last position is associated to a group hunting group, but containing a member, a Special Emergency Notification Terminal (SENT). When a call reaches the group that contains a member, the group hunting feature performs the following: • Analog phones that have been prepared for reception of an emergency notification are ordered to emit ring signal and, when going off-hook, connect to the Music-on-Hold output of a TMU board. Any ongoing call is disconnected. • Digital phones that have been prepared for reception of an emergency notification are ordered to switch to loudspeaker mode and connect to the Music-on-Hold output of a TMU board. Any ongoing call is disconnected. • IP (H.323 and SIP) extensions that have been prepared for reception of an emergency notification are ordered to display the message "Emergency Call", and are also alerted if possible. • The call is routed to the SENT, which must be configured for automatic answer and sound messages can now be transferred to analog and digital extensions comprised by the broadcast, using live voice message or recorded message.

3.4.75 EMERGENCY STATE SWITCHING

Operator Feature	
Definition	The ability for the operator to put the exchange into *Emergency State*.
Settings	Emergency switching CoS for the operator or extensions are set by the administrator.
Procedure and function	The emergency state is initiated from and indicated on the status bar. Some extensions, depending on their CoS, cannot initiate calls or receive transferred outgoing calls while the system is in the Emergency switched state. Incoming traffic is however permitted. Other extensions, operators and external lines are not affected by the emergency situation.

3.4.76 ESTIMATED WAITING TIME ANNOUNCEMENT (EWTA) FOR ACD

System Feature	
Definition	The EWTA for ACD feature provides a recorded voice announcement to the calling party when the call is queued in a busy ACD group. The announcement is based on the estimated time that the caller is likely to wait in the ACD queue. This feature works in conjunction with the Recorded Voice Announcement (RVA) feature.
Settings	EWTA is set by the administrator. See also RVA.
Procedure and function	When a call is put in the ACD queue, the caller is provided with group queue and repeat queue announcements by the RVA feature. If the EWTA feature is available, the group queue announcement can be replaced by an EWTA. The EWTA to be played is selected from up to 15 different recorded voice messages based on the estimated waiting time calculations for a particular call. The estimated waiting time ranges from 1 second to 65535 seconds.

3.4.77 EXTENDING (TRANSFER BY OPERATOR)

Operator Feature	
Definition	The ability for an operator to interconnect two parties (extending) while the two parties are connected to the operator. It is possible to extend the call before or after answer. Any external or internal traffic can be extended to an internal party (extension or operator) or an external trunk line. Extending can either be done manually or automatically.
Settings	The automatic extending is initiated by the operator. The manual extending is set from the attendant console/client or by the administrator.
Procedure and function	The following extending procedures are available: • If both parties are off-hook, the call is extended immediately when the operator presses the extend key. • If the A-party is off-hook and the B-party is on-hook, the interconnection is delayed until B-party answers. During this time the extended call is supervised. • If the B-party is an external line and the A-party is an off-hook extension, the call is extended but not supervised. **Automatic Extending**: NOW/ACS: The operator can toggle the Auto Extend on or off on the NOW/ACS. When set to on, all calls to free extensions are automatically extended. Legacy OPI terminal: The operator can toggle the Auto Extend on or off. When set to on, all calls to free extensions are automatically extended. **Manual Extending**: The operator presses the extending key on the attendant console/client to extend the call.

3.4.78 EXTENSION STATUS INDICATION

Operator Feature **Network Feature**	
Definition	The operator is notified of the extension status through information on the display.
Settings	-
Procedure and function	An operator who calls an extension is automatically informed of the status of the extension and of status changes. The display information can show, for example, if the extension is busy, busy and camped on, free (reserved), ringing, blocked or vacant.

3.4.79 EXTERNAL DATABASE (E.G. FOR ALPHA-TAGGING)

System Feature	
Definition	External database is a feature which allows connections to external LDAP databases, and it is possible to look up (query) information. The database name determines the MX-ONE functionality. There is currently one database function for public subscriber names (called “alpha-tagging”) defined.
Settings	External database(s) is initiated by the administrator.
Procedure and function	For incoming public trunk calls via e.g. ISDN or SIP trunks (or CAS trunks that convey calling party number early enough) there is an optional function to search for a public subscriber name (and later possibly other data) based on received caller number. The function uses a meta directory function (such as ESTOS) which allows several databases to be connected and queried. This functionality can be used together with the EXTERNAL DIRECTORY feature, or without that feature. The system will first check the external directory and blacklist information and then, if configured, check the external database(s).

3.4.80 EXTERNAL DIRECTORY WITH BLACKLISTING

System Feature	
Definition	External directory with blacklisting are features which allows external numbers (public subscriber numbers) to be blacklisted, i.e. not allowed to call the PBX, but also optionally to be associated with for example a name. Directories can be initiated as single numbers or ranges. Each number or range is accompanied by a number type. The customer affiliation, number and number type make an entry unique.
Settings	External directory and blacklisting of specific public subscribers is initiated by the administrator.

System Feature	
Procedure and function	For incoming public trunk calls via ISDN or SIP trunks (or CAS trunks that convey calling party number early enough), there is an optional function to check if a caller identity is blacklisted, i.e. not allowed to make calls to this PBX. The blacklisting is based on received caller number, and can be per system or per customer (tenant). An external directory can consist of the following attributes: customer number, subscriber number, subscriber number type, name, blacklist information, blacklist redirection destination number, blacklist expiration date, additional free text string. A specific Recorded Voice Announcement (vocal guidance message) exists for blacklisted calls, and is recommended to be configured. This functionality can be used together with the EXTERNAL DATABASE feature, or without the external database feature. The system will first check the external directory and blacklist information and then, if configured, check the external database(s). The external directory data has higher priority.

3.4.81 EXTRA DIRECTORY NUMBER (EDN)

User Feature	
Definition	EDNs are extra directory numbers, beside the primary number, that can be programmed on a SIP phone and are associated to a key. (See also Additional Directory Number, ADN).
Settings	Extra directory numbers are set by the administrator.
Procedure and function	An EDN works as an extra ‘telephone’ in the telephone. It is possible to both receive and initiate calls on the EDN keys. The function is available for Mitel 6900/6800/6700 SIP terminals.

3.4.82 FACILITY RESTRICTION LEVEL/TRAVELLING CLASS MARK (FRL/TCM)

System Feature **Network Feature**	
Definition	FRL/TCM is used to restrict user access to public or private network routes.
Settings	FRL/TCM are set by the administrator per user and per route.
Procedure and function	The FRL is associated as a CoS to all users and is sent as a TCM in the private network. Before an exchange permits a call attempt on the chosen route, the FRL level of the outgoing route is checked. To complete the call, the user must have an FRL higher than or equal to that of the selected outgoing route. See also the optional Priority Disconnect function, which can utilize the FRL/TCM.

3.4.83 FOLLOW-ME, EXTERNAL

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability of an extension temporarily to divert calls to users in the public networks.
Settings	The CoS is set by the administrator.
Procedure and function	An extension or the operator can order and cancel the external Follow-me diversion feature by use of a procedure. Incoming calls to the extension are then diverted (rerouted) to the external destination. However, Follow-me calls cannot be camped on to the external destination, that is, if the external destination is busy with an ongoing call, a subsequent call encounters a busy tone. Metering pulses and call information logging are registered for the extension that activated the external Follow-me.

3.4.84 FOLLOW-ME, INTERNAL

User Feature	**See also the description for *CALL DIVERSION*.**
Definition	The ability of an extension to temporarily redirect incoming calls to another extension position. The function works for internal extension or an extension within a private network using ISDN, DPNSS, SIP or H.323 signaling.
Settings	-
Procedure and function	Activated and cancelled by an authorized extension or the operator. The extension receives a special dial tone to indicate, when initiating a call, that the Follow-me state is active.

3.4.85 FORCED DISCONNECT OF OUTGOING TRUNK CALLS

User Feature	
Definition	Outgoing calls to public networks can be disconnected based on the users allowed maximum conversation time. 10 seconds before disconnect, a warning tone is given to the user.
Settings	Allowed time per user is programmed by the administrator.
Procedure and function	-

3.4.86 FORKING, MULTIPLE TERMINAL SERVICE

System Feature	**See the operational directions for *MULTIPLE TERMINAL SERVICE (PARALLEL RINGING)*.** **See also** 3.4.145 Parallel Ringing, MULTIPLE TERMINAL SERVICE on page 63.
Definition	*Forking* means registration of more than one terminal or client on the same generic extension directory number. Whether forking shall be allowed for a specific extension number is controlled by an I/O command parameter. There are a few different registration options, like normal manual log-on, and automatic log-on (which cannot be pushed out).
Settings	The Forking is set up by the administrator. Forking is only available for generic extension types. For SIP terminals, up to 4 terminals can be forked, registered on the same directory number. For other extension types, only one terminal is allowed per type.
Procedure and function	The forked extension can be ringing in parallel, similar to the parallel ringing feature, or in parallel with delay, or ringing serially, depending on configuration, but still differs from parallel ringing in some aspects due to the fact that there is only one directory number. The MX-ONE system can have interoperability with other systems using a Forking method, based on SIP extension signaling between the two systems. A user can have terminals registered as forked in both the other system and in MX-ONE, but from a user perspective the terminals appear to belong to the same system. The forking implies that all extensions belonging to a user can be reached if they are registered in either the other system or MX-ONE, regardless if the call originates from a trunk, from the other system or MX-ONE. The solution is that each user will have one number represented as a forking list in the other system, and a seizure list in MX-ONE. A call in two-party speech can be taken/moved onto another terminal belonging to the same user, by dialing a service code.

3.4.87 FREE ON SECOND ACCESS

User Feature	
Definition	On some digital or IP phones more than one call can be accepted when the function key Free on Second Access (also called Free On Busy key) is activated.
Settings	-
Procedure and function	The user has to activate the key Free on Second Access to receive calls on Line 2. Two incoming calls cannot be indicated simultaneously on the Line-keys.

3.4.88 FREE SEATING

User Feature	**See also the description for *FREE SEATING*.**
Definition	*Free seating* is a feature that allows the user to log on to any available phone, and get the user's personal categories, calls and messages. SERV parameter must be set so that Free Seating is allowed.
Settings	-
Procedure and function	For IP terminals, the user normally logs on/off using the standard log on/off procedure, but a special feature code can also be used, for terminals that do not have a log on/off procedure. Analog or digital extensions enter a special feature code to activate/deactivate free seating. An individual authorization code must be used. When logged on or activated, the user's set of function keys will be used, but only certain features will follow the user. Some features, such as group hunting, are barred. A special dial tone is used to indicate that the user is logged on/activated. Log off can also be done remotely from an operator.

3.4.89 GENERAL DEACTIVATION (GENERAL CANCELLATION)

User Feature	
Definition	The ability to deactivate all activated features such as *callback*, *follow-me*, *message diversion* and *do not disturb*, with only one procedure.
Settings	-
Procedure and function	An extension can dial a procedure that erases or deactivates all services that have been requested by or for the extension.

3.4.90 GENERIC EXTENSION

System Feature	
Definition	A Generic Extension is affiliated only to one directory number. A number of categories are affiliated to a generic extension. Those categories are collected in a number of CSPs and every generic extension must be affiliated to a CSP. All extensions except analog, CAS, digital extensions and ISDN-S0 are generic extensions.
Settings	A generic extension is initiated by the administrator.
Procedure and function	After the basic setup has been initiated for the directory number, different applications can be defined to affiliate this directory number to an extension position.

3.4.91 GROUP HUNTING

User Feature	**See also the description for *EXTENSION GROUPS*.**
Definition	A group of extensions can be called with a common number. If all extensions are busy, incoming calls can queue until a user becomes free. There are two variants of hunt groups; one with sequential alerting of members (ordinary hunt group, one member alerted at a time), and one (cascade ring group, allowing fewer members) with parallel alerting of the members.
Settings	Hunt group configuration data settings and the allocation of members is done by the administrator.
Procedure and function	A group of extension can be called with a 2–10 digit group pilot number. Incoming calls are routed to a free extension member in the group, either with sequential hunting or evenly distributed, or in parallel (for cascade ring group). All extensions in a group keep their own private numbers and CoS codes. An extension can be a member of several (up to 4) groups. An extension can temporarily withdraw from the group by either logging out of a group or by activating Follow-me to its own phone. There are also group log-on/log-off procedures. The members can also be temporarily withdrawn from the group if they are alerted but do not answer. Calls to a group from which all members have excluded themselves are diverted to the group's divertee position.

3.4.92 HLR BACKUP/HLR REDUNDANCY

User Feature	**See also the description for *HOME LOCATION REGISTER REDUNDANCY*.**
Definition	HLR backup/HLR Redundancy is a feature which means that IP extensions can, on certain conditions, register to a backup Home Location Register in another server than the ordinary HLR.
Settings	Normally none, but the service can be turned off by the system administrator. Default is that the service is not active.
Procedure and function	HLR backup/HLR Redundancy is available to IP terminals, that is, H.323 and SIP telephones/clients, and means that when an arbitrary Server (LIM) with HLRs becomes unavailable, these extension types can register or re-register in any Server (LIM) in operation in the same system. In other words, a backup HLR can be created and the extension can be registered to that backup HLR. When the ordinary HLR recovers, it is possible to re-register to that HLR. This is however not done immediately, but with a delay. Note that several services, like group functions and queue functions, are lost while registered to a backup HLR. There is no specific user procedure. See also Redundancy, server and network.

3.4.93 HOLD/PARKING WITH COMMON CALL PICK-UP

User Feature	
Definition	The ability for a user with a digital, analog, or IP phone (SIP or H.323) phone to park a call that can be picked-up from specific phones.
Settings	For digital extensions the feature requires an MDN-key. For IP phones, the feature requires an MNS key.
Procedure and function	To park on digital extensions or IP phones, the user presses the key where the call appears. The parked call can be picked up from any other extension with an MDN-key or (for IP phones) MNS-key that represents the extension putting the call on hold.

3.4.94 HOLD/PARKING WITH INDIVIDUAL CALL PICK-UP

User Feature	
Definition	The ability to park a call and pick it up from any ATS, RXN, DECT, or SIP terminal, that is from all single line access extensions.
Settings	This function is only available if the SIP extension is configured as single line access.
Procedure and function	The extension uses the procedure for Inquiry and goes on-hook to park the call. The call can be picked up from any other phone by doing an individual call pickup (that is, call your own number, get busy message, and request pickup using suffix digit). If the call still is parked after 30 seconds, the extension’s phone is recalled.

3.4.95 HOLD/PARKING USING CALL PARK POOL AND INDIVIDUAL CALL PICK-UP

User Feature	
Definition	The ability to park a call ‘remotely’, at a dedicated Call Park Pool (CPP), i.e. a special hunt group, and pick it up from any extension in the same system that can use individual call pick-up.
Settings	This function is only available if the involved hunt group and SIP extensions are properly configured.
Procedure and function	The extension uses the procedure for Hold/Parking/Inquiry, i.e. parks the original call, makes a new call to a dedicated Call Park Pool hunt group number, and makes a transfer to park the call. The call disappears from the transferring/parking party’s telephone. Announce the call to the potential picking parties. The transferring/parking party is free to make or receive new calls. The call can be picked up from any other phone (in the same system) by doing an individual call pickup (that is, call the CPP member number, get busy message, and request pickup using suffix digit). If the call still is parked after 5 minutes, the parking party’s extension is recalled (if it is a generic extension).

3.4.96 HOSPITALITY

User Feature **System Feature**	**See also the description for *HOSPITALITY APPLICATION*.**
Definition	Hospitality Application: Offers functionality specially aimed for the Hospitality industry: Hotels, Hospitals, Cruise ships, Exhibition Centers and Convention Centers.
Settings	Set by the administrator.
Procedure and function	The hospitality function includes special classes of service for extensions, special display messages, and billing functions. There can be 4 types of Hospitality classes of service: • Normal extension • Room vacant • Room occupied • Service quarter. Room vacant/occupied class of service is assigned to the guest room extensions. Service quarter class of service is assigned to extensions used by the hospitality staff to get the information of guests. Service quarter has special additional information string that can be displayed on it. This text string has information about guests.

3.4.97 HOT LINE, DELAYED

User Feature	
Definition	The ability of an extension to be connected automatically to a predefined position after the handset has been lifted for a certain time.
Settings	The connection position (address) is set per extension by the administrator. The delay time can be modified by the administrator.
Procedure and function	Ringing signals are sent to the preprogrammed party (another extension, operator, speed dialing number) a certain time after the extension lifts the handset. Before that time, the extension can dial as any normal extension.

3.4.98 HOT LINE, DIRECT

User Feature	
Definition	The ability of an extension to be automatically connected to a predefined position immediately after lifting the handset.
Settings	The connection position (address) is set per extension by the administrator.
Procedure and function	Ringing signals are sent to the preprogrammed party (another extension, operator, speed dialing number) when the extension lifts the handset.

3.4.99 IMMEDIATE SPEECH CONNECTION (AUTOMATIC CALL ACCEPTANCE)

Operator Feature **User Feature**	
Definition	The ability to answer incoming calls without having to press the answer key. Automatic answer is also available for some extension types with authority to use it.
Settings	The extension CoS is set by administrator. Immediate speech connection is initiated from the attendant consoles/clients or by the administrator.
Procedure and function	The attendant can toggle the feature on or off on the attendant console/client. Voice contact from the incoming call is thereby automatically made with the attendant.

3.4.100 INCOMING AUTOMATIC INTER-PBX CALLS

User Feature **Network Feature**	
Definition	Incoming calls from other PBXes can be routed directly to an extension.
Settings	Acceptance of automatic direct calls is set per route by the administrator.
Procedure and function	The route can be programmed to accept calls from other private exchanges. Automatic calls from other private exchanges are processed as Direct In-Dialing calls from the public network. **NOTE:** The call can be rerouted (for example, to an operator that gets information about the dialed number and reason for rerouting) if congestion occurs, or if the extension is not answered, busy, vacant, blocked for direct in-dialing or in line-lockout state.

3.4.101 INCOMING CALLS VIA OPERATOR

Network Feature	
Definition	Incoming calls from the PSTN or other PBXes can be routed to the operator.
Settings	Set by the administrator.
Procedure and function	The operator receives the call and can extend it to any extension in the private network.

3.4.102 INFORMATION SYSTEM/INTERCEPTION SERVICE (ICS)

User Feature **System Feature**	**See also the description for *INTERCEPTION SERVICE*.**
Definition	An information system can consist of, for example, a message switching system of type interception computer, or voice mail system. The MX-ONE uses a standardized interface (GICI) between the exchange and the interception computer, where the systems can exchange information. The feature *message diversion* is activated by an extension procedure containing an interception message. The interception message is sent to the connected interception computer. All new calls to an extension which has activated interception diversion are diverted directly to an answering position programmed as the Message Diversion position. The purpose is to provide answering position personnel with a better means of giving callers meaningful interception messages. The features *message diversion* and *message waiting* are related to this feature.
Settings	The information system/interception service is set up by the administrator.
Procedure and function	The procedure for booking message diversion includes an interception message which is sent direct to the interception computer.

3.4.103 INQUIRY

User Feature	**See also the description for *THREE-PARTY SERVICES*.**
Definition	The ability of an extension to park a call and make an inquiry.
Settings	-
Procedure and function	The original call (internal or external) is parked by the extension using the procedure for inquiry. The required number (internal, external or operator) is dialed. The parked party cannot overhear the inquiry call. Referral to the parked party is done by using the refer back procedure.

3.4.104 INSTANT MESSAGING CALL (USING MSRP)

User Feature	
Definition	The ability of SIP extension type of terminals/clients with instant messaging support, to make and receive ‘instant messaging calls’, i.e. calls with a text string of limited length as ‘media payload’. See RFC 4975, Message Session Relay Protocol, for more information. MSRP could also be conveyed through the private or public homogeneous SIP network.
Settings	There is a CoS for instant messaging support for the Personal Number and Bearer Capability of a SIP route, which is set by the administrator.
Procedure and function	An instant messaging call is initiated by a SIP extension. For instant messaging calls that encounter certain services, like Diversion, Personal Number deflect, Conference, Parking, Transfer and others, the service will be bypassed/rejected.

3.4.105 INTERCOM

User Feature	
Definition	The ability of SIP extensions to have an intercom function, i.e. to push a single key to be connected to a pre-defined extension party, and to activate the loud-speaker on the called party immediately.
Settings	The Intercom function must be configured using the EDN, Hotline and Immediate/Automatic answer features, which are all set by the administrator.
Procedure and function	An intercom connection is established by pushing a single line key, and the SIP extension at the pre-defined destination number, which should be an EDN (SIP extension extra directory number), will then be immediately connected, with its loud-speaker active, if appropriately configured. The called party will be muted until it actively connects itself by pushing the concerned line key. The called party should preferably have a similar configuration, able to set up an Intercom connection in the opposite direction. Note: DTS/ADN can have a similar function, but without muting, so it is a Hotline function rather than Intercom.

3.4.106 INTRUSION ON BUSY EXTENSION OR EXTERNAL LINE

Operator Feature User Feature Network Feature	
Definition	The ability of the operator or extension user to intrude on the conversation of an engaged extension (which does not have other free lines).
Settings	Both the intruder and the intruded party need specific CoS (capability and protection levels respectively) set by the administrator.
Procedure and function	A suffix digit is used to enter the ongoing conversation. An intrusion tone is heard by all parties, which are connected in a kind of three-party conference connection. The function works for operator (as intruding party) internal extension or an extension within a private network using ISDN, DPNSS, SIP or H.323 signaling.

3.4.107 INTRUSION WITH FORCED RELEASE

Network Feature Operator Feature	
Definition	The ability of an attendant, after intrusion, to force release of the third party.
Settings	Intruded party need specific CoS (protection level) set by the administrator. The intruder operator always has highest capability level as default.

Network Feature **Operator Feature**	
Procedure and function	The attendant can force release the third party by pressing the Intrusion key once more. Forced release cannot be initiated earlier than one second after intrusion. The function works internally and within a private network using ISDN, DPNSS, SIP or H.323 signaling.

3.4.108 IP EXTENSION (SIP AND H.323)

User Feature	**See also the descriptions for *SIP EXTENSION* and for *IP (H.323) EXTENSION*.**
Definition	The IP extension is a type of terminals that is connected to the IP network. The *IP extension* feature allows the connection to the MX-ONE Service Node of terminals that are compliant with either the ITU-T H.323 or IETF SIP standards (RFCs).
Settings	The IP extensions and their CoS are initiated by the administrator. A ‘forced gateway’ CoS can be set by the administrator.
Procedure and function	The H.323 extensions comply with relevant parts of the ITU-T H.323 standards, plus some proprietary additions. The SIP extensions comply with IETF RFCs for the Session Initiation Protocol, plus some proprietary additions. The functionality of the two protocols is similar, but there are differences. See for example the MIVOICE MX-ONE FEATURE MATRIX or the description

3.4.109 IP NETWORKING

Network Feature	**See also the description for *IP NETWORKING*.**
Definition	The *IP networking* feature allows multimedia communications between an MX-ONE and other H.323/SIP compliant systems, making use of the IP network to transmit the media. The IP network uses TCP/IP as the underlying protocol and therefore the voice is converted into packets to be transmitted through the IP network, and then unpacked in the other end.
Settings	The *IP networking* feature is set up by the administrator.
Procedure and function	*IP networking* allows that an MX-ONE can be connected through the data network to remote PBXes in multi-site organizations. The IP network uses TCP/IP as the underlying protocol and therefore the voice is converted into packets to be transmitted through the IP network, and then unpacked in the other end. The use of the IP network to transmit voice may reduce transmission costs, and allows the transmission of different media (voice, video, data). With the IP networking feature is a complete integration of voice and data communications obtained. Therefore, is only one network needed (the data network).

3.4.110 ISDN FALLBACK FOR VOIP MEDIA

System feature **Network Feature**	
Definition	The ability to route media for VoIP calls, within one PBX, via the public ISDN, as an alternative to RTP via TCP/IP (or to Group Switch) connection. It can also be used as fallback when encountering Call Admission Control congestion for the IP connection between media gateways and Servers in the same PBX system.
Settings	Inter gateway routing for VoIP media via public ISDN is configured by the administrator. Appropriate ISDN routes must exist.
Procedure and function	The system uses TCP/IP connections between the different Servers (LIMs), but if public ISDN is available, and the inter gateway routing feature is configured, it is possible to route the media connections via ISDN instead of via TCP/IP. Dedicated virtual routes are defined and used for the feature. The ISDN signaling must support either UUI (proprietary data) or transparent calling and called party numbers, in the SETUP message.

3.4.111 ISDN QSIG NETWORKING

Network Feature	**See also the description for *NETWORKING*.**
Definition	ISDN networking allows communication between the MX-ONE and other PBXes using QSIG signaling. The connection requires ISDN E1/T1 connections between the systems to be connected.
Settings	The ISDN networking is set up by the administrator.
Procedure and function	ISDN networking allows that a telephony system can be connected through leased lines to remote PBXes in multi site organizations or to PBXes of other organizations.

3.4.112 ISDN S0 TERMINAL INTERFACE

User Feature	
Definition	The ISDN S0 terminal interface, also called a Basic Rate Access (BRA) (2B+D) interface, is an interface to which ISDN terminals can be connected. The ISDN S0 terminal interface has two 64 kbits/s B-channels, and two equipment positions are needed for one interface. The ISDN S0 terminal interface allows voice and data transmission on ordinary phone lines. An ISDN S0 terminal can be any terminal equipment with an S0 interface or a terminal adapter with a connected non-ISDN terminal, for example, a telefax GP4, PC with ISDN board, PC with ISDN board and a phone, terminal adapter with a handset, video phone, Local Area Network, LAN, connection or an ISDN phone.
Settings	ISDN S0 extensions are set by the administrator.

User Feature	
Procedure and function	-

3.4.113 LAST NUMBER REDIAL

User Feature	
Definition	There are two types of “centralized” redial features: **Stored Number Redial:** The ability for a user to redial the last stored number by pressing a specific key. The last stored number can either be an internal number or an external number. Only for digital extensions. **Last Number Redial:** Dialed numbers are automatically stored and can be re-transmitted by the extension using a service code. This feature can be configured to apply only to external numbers, or to all numbers.
Settings	The administrator selects via a system parameter, if the last number redial is valid only for public calls or for both private network calls and public calls, or for all calls.
Procedure and function	All dialed numbers from each extension are stored. A stored number is erased when a new number is dialed. The extension uses a certain code to re-transmit the stored number. Some extensions may also have a similar local function in the terminal. Note: This describes the LNR numbers stored by the MX-ONE Service Node. Certain terminal types (DECT, SIP, H.323...) also have local similar functions in the telephones.

3.4.114 LEAST COST ROUTING

System Feature	**See also the description for *LEAST COST ROUTING*.**
Definition	*Least Cost Routing* allows the system to select the most economical route for an outgoing public call. Least Cost Routing performs analysis of the dialed number including the Least Cost Routing access code and attempts to route the call over the most economic available route at any time based on the following: • external line availability • user's Routing CoS • user's Trunk Call Discrimination CoS. **Time of Day** A function that allows the system to make the selection of the most economical route for an outgoing call depending on the time of day and the weekday. Thereby it is always possible to select the most economical route even if the cost relations between the different routes vary with the time of day and the weekdays. **On/Off Hook Queuing** Off hook queuing is initiated automatically by the PBX, while on hook queuing is invoked manually, like Callback. **Transit Network Selection** **Expensive route warning tone** A special tone is returned to the calling party when a call is overflowed to an alternative route which is marked as expensive.
Settings	The Least Cost Routing tables are set up by the administrator.
Procedure and function	-

3.4.115 LICENSE USAGE REPORTS

System Feature	
Definition	The feature License Usage Reports provides an easy way to keep track of how licensed resources are utilized over time. Daily peak values and snapshots of currently seized licenses are periodically taken and archived and distributed via e-mail. Default distribution interval for license usage report is once a month.
Settings	The feature is controlled by a license, making the feature mandatory in Cloud configurations, but optional in CPE configurations.
Procedure and function	-

3.4.116 LOG-OFF RESTRICTION

System Feature	
Definition	Log-off restriction is a feature which restricts the log-off possibility on Mitel SIP extensions. They can be fully restricted, where log-off is not allowed at all, or semi-restricted, which means the first registration will be considered the “default extension”. The log-off key will then be labeled “TempUser”, allowing temporary registration to own extension. An automatic registration back to the default extension will take place if the log-off key is pressed, or after 4 hours.
Settings	The extension CoS is set by the administrator.
Procedure and function	The log-off key will not function, or will function with restrictions, if log-off restrictions are configured.

3.4.117 MALICIOUS CALL TRACING

User Feature Operator Feature	
Definition	The ability of an extension or attendant, which is or has been participating in an incoming public trunk call where the trunk supports MCT, to request tracing of a malicious call.
Settings	The extension CoS is set by the administrator. The attendant has no CoS.
Procedure and function	For an analog, remote/mobile extension or SIP extension, the feature is requested by putting the call on hold, and then dialing a procedure. The request to trace is sent to the public network which performs the actual tracing. For a digital extension or H.323 terminal, a dedicated key is required for tracing. For an operator, a specific dialing procedure is used.

3.4.118 MANAGER-SECRETARY

User Feature	
Definition	This is not a feature in it's own right, it is usually provided via a combination of existing features like diversion, Personal Number, monitoring (MDN/MNS) and diversion by-pass. Combinations of these features provide secretarial screening by one or more secretaries for one or several managers. The feature called Boss-secretary is a special version of manager secretary functionality.
Settings	Initiated by the administrator.
Procedure and function	See Boss-Secretary and the respective features for details

3.4.119 MANUAL ANSWER

Operator Feature	
Definition	The ability to accept incoming calls manually. See also *Immediate speech connection (automatic call acceptance*).
Settings	Set from the attendant console/client or by the administrator.
Procedure and function	The attendant presses the answer/extend key on the attendant console/client to answer the call.

3.4.120 MANUAL MESSAGE WAITING (MMW)

User Feature Network Feature	
Definition	The ability of an extension or operator to manually notify an extension that a message is waiting.
Settings	An extension CoS is set by the administrator.
Procedure and function	The feature is requested by dialing a specific procedure or pressing the MMW key. The destination extension then switches on the message waiting indication. When the extension which received the indication becomes aware that there is a waiting message, a call can automatically be set up to the party who requested the MMW indication.

3.4.121 MASTER EXTENSION

User Feature	
Definition	An extension which is permitted to order the group do not disturb, bypass of an extension with group do not disturb activated or follow me for an internal group hunting group by dialing predefined procedures. Not valid for generic extension.
Settings	The master extension is initiated by the administrator.
Procedure and function	-

3.4.122 MAXIMUM CALL DURATION

User Feature	
Definition	The maximum duration time of an outgoing call to PSTN can be set for analog, CAS, digital, and generic extensions.

User Feature	
Settings	The maximum call duration is initiated by the administrator.
Procedure and function	-

3.4.123 MESSAGE DIVERSION

User Feature	For more information, see the description for *INTERCEPTION SERVICE*.
Definition	The ability of a user when absent, to set the reason for absence and the expected return time or date. Callers to the extension with active message diversion are informed either directly on the digital phone or IP phone display, or the caller is routed to an answering position where the message diversion information can be read. The answering position can either be an individual or a common answering position set for the user with active message diversion. Typical answer positions are an operator or a secretary.
Settings	Users must have an active individual or common diversion answering position. The meaning and syntax of the Message Diversion messages can be changed by the administrator.
Procedure and function	Message Diversion is activated by a procedure from the phone. The operator can assist in setting up the message diversion feature for any user. The actual absence reasons can be set and changed per system, but they have a default setting per application system and language. One digit 0-9 represents an absence reason. For example the Standard application system has the following default settings for English. In the procedure *FC*Reason*(Time/Date)#, dd= day, hh = hour, mm = month/minute: Reason*Time/Date Lunch=0*hhmm Busy=1 Absent=2*hhmm Meeting=3*hhmm Trip=4*mmdd Course=5*mmdd Vacation=6*mmdd DayOff=7*mmdd GoneHome=8*hhmm Illness=9*mmdd

3.4.124 MESSAGE WAITING

System and User Feature	For more information, see the descriptions for *MESSAGE WAITING, INTERCEPTION SERVICE* and *CSTA PHASE III.*
Definition	A feature for notifying an extension of messages that have not yet been read, heard or recorded and which are stored in one or several information systems. An information system can consist of, for example, a message switching system of type interception computer, or voice mail system. The MX-ONE uses standardized interfaces (GICI, CSTA Phase III) between the exchange and the interception computer, where the systems can exchange information.

System and User Feature	**For more information, see the descriptions for *MESSAGE WAITING, INTERCEPTION SERVICE* and *CSTA PHASE III.***
Settings	The message waiting feature is set up by the administrator.
Procedure and function	The *message waiting* feature can be used by several information systems connected in parallel to the system. An information system can be in the form of an interception computer, a voice mail system, or a text message system. The feature can be introduced even if not all information systems connected are capable of handling the signaling required for Message Waiting. When a message has been registered in an information system, it is signaled to the MX-ONE, which then notifies the relevant extension. After a message has been presented to the receiving party or canceled by other means, the message system informs the sending party and the notification ceases unless other messages to the extension exist.

3.4.125 MESSAGE WAITING INDICATION (MWI)

User Feature	
Definition	The ability of an extension to receive a Message Waiting Indication (MWI) in the form of a LED, a display message, or a special dial tone at Off-hook, depending on type of extension.
Settings	An extension CoS is set by the administrator.
Procedure and function	The feature is requested by an interception computer system or can be set manually for a particular extension when there is at least one message waiting for that extension user.

3.4.126 MF SIGNALING

System Feature	
Definition	The ability to communicate with PSTN or other PBXes, using Multi Frequency (MF) signaling over analog trunks or digital CAS trunks. The tone signals are created by combining two frequencies from a group of 6 frequencies within the frequency band 300 - 3400 Hz.
Settings	The protocols for MF signaling are set by the administrator.
Procedure and function	-

3.4.127 MICOLLAB ADVANCED MESSAGING

System Feature	**For more information, see the operational directions for *MITEL MiCollab Advanced Messaging VOICE MAIL* and the administrator's guide for *OPENTEXT RIGHTFAX.***
Definition	Mitel MiCollab Advanced Messaging (former OneBox) consists of Voice Mail and Fax Mail. An optional feature for integration with e-mail platforms offers full Unified Messaging capability. **Voice Mail:** Allows callers to leave voice messages for users that are not available to answer a call. Notification services, such as SMS, E-mail and MWI are provided to alert the user of the arrival of new voice mails. Mitel MiCollab Advanced Messaging is connected through the IP network as a SIP extension connection to MX-ONE. Additionally, IP connections are created to enable MWI notification and Message Diversion integration. The system comes with full auto-attendant functionality in a variety of languages to handle inbound call processing. Furthermore, the Mitel MiCollab Advanced Messaging system can be integrated with all major e-mail platforms to provide full Unified Messaging services. Users are provided with a variety of interfaces to access their voice mail box (Fixed phone, Mobile, e-mail or web page) to listen to and manage their voice mails. **Fax Mail:** Allows users to create, send, receive, and manage faxes directly from a desktop computer. The fax can be enhance by adding a cover page, overlaying a form, attaching documents, and inserting graphics. FaxUtil is the RightFax mailbox where users create, view, print, and manage faxes. With FaxUtil, users can forward, route, and delete faxes. They can view other users' fax mailboxes and delegate views into their personal fax mailboxes. RightFax also supports connectors for integration with e-mail platforms so users can manage their faxes directly from their e-mail inbox.
Settings	Calls to be routed to the voice mail system is set up by the administrator. The signaling connection(s) and the (SIP trunk or SIP extension) media channels between the MX-ONE system and the MiCollab Advanced Messaging system must be configured by the administrator.
Procedure and function	**Voice Mail:** The user has several option to access their voice mailbox to retrieve messages. The most common is to access the TUI through the phone using menu driven option procedure. Alternatively, the user can access the Voice Mailbox using a Web based user interface that will offer them a variety of option to manage their voice mails and mailbox options. Finally, it is also possible to integrate the Mitel MiCollab Advanced Messaging system to the corporate e-mail server to have full unified communications. This means the user can receive, listen to and mange their voice mails directly from their normal e-mail in-box. **Fax Mail**: Create the call processor mailbox and specify a name that identifies the source of the fax, such as the Fax Server. Inbound faxes will be routed to the faxmail server and based on the called party number will be stored in user's fax mailbox. If the e-mail integration is present, the fax notification will be sent to the user's e-mail inbox enabling them to access and manage their fax mails directly from their e-mail account.

3.4.128 MONITORING KEY (MNS)

User Feature	
Definition	A monitoring key (also called MNS) is a key on a digital phone or an IP-phone that can be used for monitoring of another phone. It is possible to monitor a directory number on digital, analog, IP and remote/mobile extensions. The directory numbers that are monitored can be represented on several IP or digital phones.
Settings	The monitoring keys has to be programmed by the administrator.
Procedure and function	The LED on the monitoring key indicates the traffic status of the multiple represented number. The user can pick up a call on a Monitoring key by pressing the key when the LED is flashing indicating an incoming call on the monitored number. When the key is idle the user can by pressing the key initiate a call to the monitored number. For IP terminals, only MX-ONE IP phones support programmable keys which can be used as Monitoring keys. The MNS key can be used to build up boss-secretary groups. See also Boss-Secretary and Multiple Represented Directory Number (MDN).

3.4.129 MULTI-DIRECTORY DIVERSION/DO-NOT-DISTURB FOR DTS

User Feature	
Definition	The ability to activate or deactivate follow me, direct diversion, message diversion or do not disturb for both the primary directory number (ODN), as well as the Additional Directory Numbers (ADNs) on the same digital phone.
Settings	Set by the administrator.
Procedure and function	If the extension's ODN has the MDD setting, there is no specific procedure for the user. The ordinary re-direction/DND procedures are used, but will affect all ADNs and ODN.

3.4.130 MULTI-DIRECTORY DIVERSION/DO-NOT-DISTURB FOR SIP PHONES

User Feature	
Definition	The ability to activate or deactivate follow me, external follow me, message diversion or do not disturb for both the primary directory number (ODN), as well as the Extra Directory Numbers (EDNs) on the same SIP terminal (Mitel 6900/6800/6700 models only).
Settings	Set by the administrator.
Procedure and function	If the extension's ODN has the MDD setting, there is no specific procedure for the user. The ordinary re-direction/DND procedures are used, but will affect all EDNs and ODN. There are functional differences from the MDD on DTS.

3.4.131 MULTI MEMBER BUSY ON PRIMARY NUMBER OR ADDITIONAL NUMBER (ADN)

User Feature	
Definition	If Multi Member Busy (MMB) is set on the primary directory number and if the ADN is busy, then the primary number is also treated as busy for any incoming calls. If the additional directory number has MMB set and if the primary number is busy, then the additional directory number is treated as busy for incoming calls.
Settings	MMB is set by the administrator.
Procedure and function	-

3.4.132 MULTIPLE ACCESS LINE

User Feature	
Definition	The Triple Access Line is a set of three (or more) keys on digital and IP phones: • Line 1 • Line 2 • ... • Inquiry
Settings	-
Procedure and function	Any of these line keys can be used for outgoing calls. For incoming calls Inquiry key cannot be used. The number of lines is terminal dependent.

3.4.133 MULTIPLE REPRESENTED DIRECTORY NUMBER (MDN) AND SHARED CALL APPEARANCE

User Feature	
Definition	The ability to multiple represent a directory number provides a feature which is also called *specific line pick-up* or Shared Call Appearance (SIP phones). This feature makes it possible to answer (pick up) or make calls on behalf of the phone which is multiple represented. The difference between MDN/SCA and the MNS feature is that when a user presses an MDN/SCA key, the extension number being monitored is occupied by the user. When pressing an MNS key, a call being presented on the monitored extension number will be picked up and the monitored extension number becomes free.
Settings	The MDN/SCA feature key and which key to assign the MDN/SCA to is set by the administrator. SCA with or without bridging (conference) can be set by administrator per SCA number.

User Feature	
Procedure and function	**MDN:** A directory number on another extension can be multiple represented with the traffic feature *specific line pick-up*, in one or more digital extensions. Each MDN is assigned to an optional function key which becomes an MDN key. The LED indicates the traffic status of the represented number. The MDN can be used to build up boss-secretary groups. See also *Monitoring Key (MNS)*. **SCA:** A directory number on another extension can be multiple represented with the traffic feature *Shared Call Appearance*, in one or more SIP extensions. Each Shared Call Appearance line is assigned to an optional line key, soft key or function key which becomes an SCA key. The LED indicates the traffic status of the represented number. The SCA can be used to configure for example office work groups.

3.4.134 MUSIC ON HOLD/WAIT

System Feature	See also the description for *RECORDED VOICE ANNOUNCEMENT*.
Definition	A feature for providing sound, such as music or marketing information, for example, to a parked or queued subscriber or extension. The sound files are either stored in memory, or streamed. Music on Hold/Wait is not available to operators or for calls queued to operator. The feature is available for incoming calls on a trunk as well as for internal calls.
Settings	The Recorded Voice Announcement feature is used for Music on Hold/Wait. There are two supported configurations, either with stored wav-files or MP3-files or with streaming via SIP. Settings can be separate for trunk calls and internal calls.
Procedure and function	If initiated by the administrator, the MoH/MoW (which is a kind of recorded voice announcement) can be played back automatically, for some selected traffic cases, such as parked calls or calls queued towards for example a group.

3.4.135 NAME AND NUMBER LOG

User Feature	See also the description for *NAME AND NUMBER LOG, CENTRALIZED*.
Definition	*Name and number log* is a software feature for logging calls (on SIP phones or digital extensions). Both made/outgoing and received/incoming calls can be logged. The received calls can be classified as “answered” or “unanswered/missed”. The log stores number and name, date, time and certain additional service related information.
Settings	For SIP extensions: Whether the functions shall be active for a specific extension service profile can be set by the administrator. Terminal configuration files must also be updated to select if central or local log shall be used. For DTS: Programmable keys with a LED (so called soft keys) can be set by the administrator. An option to only log *unanswered* calls can also be set by the administrator.

User Feature	**See also the description for *NAME AND NUMBER LOG, CENTRALIZED*.**
Procedure and function	By using a SIP extension (Mitel 6900/6800 models), with the centralized name and number log function supported and active, the user can display, dial and delete the logged calls. The actual interface is controlled by the terminal, and may be for example menu or key based. By using a soft-key on a DBC 223 or DBC 225 digital phone, the user can display, dial and delete the stored numbers on the logged calls. **Note:** Other extension types and the Mitel 6900/6800 SIP phones may have similar local functions in the terminal, but this feature only refers to the centralized call log functionality (where the log is stored in the PBX).

3.4.136 NIGHT BELL (UNIVERSAL NIGHT SERVICE)

System Feature	
Definition	Incoming calls are signaled on a common alerting system, for example, bells. Any extension can answer such calls by dialing a predetermined answering procedure.
Settings	Set by the administrator.
Procedure and function	An incoming external call to a PBX switched for night service can be diverted to a night service positions where it is queued. There can be a maximum of 30 night service positions in the PBX. All the signaling devices which are associated to the night service positions are activated and any extension in the PBX can access the call by dialing a predetermined code.

3.4.137 NIGHT SERVICE

System Feature	
Definition	Incoming calls during the night are routed to preprogrammed answering positions.
Settings	Type of Night Service and the assignment of answering positions are set by the administrator.
Procedure and function	Night Service is activated by the operator or automatically by a programmer time. See also Night Bell (Universal Night Service).

3.4.138 NUMBER ANALYSIS/NUMBER SERIES

System Feature	**See also operational directions for *NUMBERING*.**
Definition	The ability to analyze dialed or received numbers and feature codes, including separators.
Settings	Number series are initiated and changed by the administrator.

System Feature	See also operational directions for *NUMBERING.*
Procedure and function	When a user makes a call, or when certain features are executed, the dialed or received digits or characters are analyzed regarding number type, length, and range. Internal number length (extension, group and attendant directory numbers) is maximum 10 digits. External, public and Least Cost Routing destinations allow a maximum of 5 digits. Each customer can have its own number analysis.

3.4.139 NUMBER CONVERSION AND BEARER CAPABILITY SUBSTITUTION

System Feature	See also operational directions for *NUMBER CONVERSION AND BEARER CAPABILITY SUBSTITUTION.*
Definition	*Number conversion* and *bearer capability substitution* are features that perform conversion of sent and received numbers, bearer capabilities and teleservices.
Settings	The database contents and route CoS are initiated by the administrator.
Procedure and function	The system automatically performs number conversion or bearer capability substitution for the calls that need these services. The primary application of this feature is to identify that an incoming public number corresponds to a remote/mobile extension. ISDN Type Of Number is used to determine how the conversion shall be done.

3.4.140 OPERATOR/ATTENDANT

Operator Feature	See also operational directions for *PBX OPERATOR TRAFFIC.*
Definition	The attendant has the ability to control the following from the console/client: • automatic or manual answer • automatic or manual extending • continuous tone on call or tone burst call • tone ringer character • language
Settings	Features in the MX-ONE for the operator is set from the attendant console/client, or by the administrator.
Procedure and function	-

3.4.141 ORIGINAL A-NUMBER

System Feature	**See also the description for *ORIGINAL A-NUMBER*.**
Definition	The Original A-number feature allows the transferred, extended or diverted to-party (the C-party) to see or retrieve the originating calling party's (the A-party's) number in the following cases: • The B-party has external follow me to the C-party activated • The C-party is an answer position in the B-party's active list for personal number • The B-party has transferred to C-party Note that the public network may not accept the original A-number as CLI for the call and replace the CLI with the main number of the DID series. DivLeg2 is a subset of the ETSI call forwarding standard that will send the original A-number in a special information element that will be shown to the user. MX-ONE supports DivLeg2.
Settings	The CoS is set by the administrator.
Procedure and function	-

3.4.142 OUTGOING AUTOMATIC CALL

User Feature	
Definition	The ability of an extension to make outgoing calls without assistance from the operator.
Settings	The CoS is set by the administrator.
Procedure and function	The extension dials the route access code and normally, on receipt of a dial tone, the wanted external number. A supervisory tone is sent after the route access code if the extension is barred from making outgoing calls or using the route access code.

3.4.143 OUTGOING CALL VIA OPERATOR

User Feature **Network Feature**	
Definition	The ability of an extension to make an outgoing call with assistance from the operator.
Settings	The CoS is set by the administrator.

User Feature **Network Feature**	
Procedure and function	For this traffic case, three sub-cases can apply: • The extension replaces the handset after ordering the call. The extension dials the operator access code. After ordering the call, the extension replaces the handset. The operator dials the required number, awaits answer, calls the extension, awaits answer, and finally extends the call. • The extension does not replace the handset after ordering the call. The operator dials the required number and extends the call without the extension having to replace the handset. • The operator extends the call with dial tone. The operator dials the route access code, awaits dial tone from the called exchange, and extends the call, that is, allows the extension to dial the actual destination.

3.4.144 PAGING

User Feature	
Definition	*Paging* is used to send one-way messages to persons with either a pager or with a personal code over an acoustical/optical system. It enables persons to be paged from an extension, an operator or external line.
Settings	Directory numbers and codes for paging are set by the administrator.
Procedure and function	The person or group of persons are paged by using the directory numbers. The directory number can be an ordinary extension number, a specific paging number or a common directory number for a group of persons who are to be paged simultaneously. Directory numbers can also be linked to an arbitrary paging code (for example, radio receiver number). *Paging* can be initiated through a specific procedure or automatically during call transfer.

3.4.145 PARALLEL RINGING, MULTIPLE TERMINAL SERVICE

User Feature	**See the operational directions for *MULTIPLE TERMINAL SERVICE (PARALLEL RINGING)***
Definition	The ability for a user to have three voice extensions (grouped as a parallel ringing list) that ring simultaneously when the user receives an incoming call. The incoming call can be answered by any of the extensions.
Settings	Parallel ringing lists are set by the administrator.

User Feature	See the operational directions for *MULTIPLE TERMINAL SERVICE (PARALLEL RINGING)*
Procedure and function	Parallel ringing has the following characteristics: • A call to a parallel ringing list is made through the main extension number. • A call made to a secondary extension on the parallel ringing list, will not initiate parallel ringing. • An extension cannot be defined in more than one parallel ringing list. • When a call is made to the parallel ringing list it is possible to transfer the call to a secondary extension, or to the main extension.

3.4.146 PARKING AND RETRIEVAL OF PARKED CALLS

Operator Feature	
Definition	The ability for the Operator to put calls on hold while answering other calls.
Settings	This feature is programmed by the administrator.
Procedure and function	The Operator has three methods available to put a call on hold: • If the extend key is used the call is put on hold for 30 seconds and is then signalled as a recall. • Use of one of the supervision links. A call parked on a supervision link can be retrieved at any time. • Use of the monitoring (listening) link. A call parked on the monitoring link can be retrieved at any time.

3.4.147 PARKING FROM AN EXTENSION

User Feature	
Definition	The ability for the user to put calls temporarily on hold while answering other calls.
Settings	This feature is programmed by the administrator.
Procedure and function	Press the line-key to park the call and then replace the handset. To re-admit the call press the line-key where the call is parked. The speech connection is now re-established. The procedure is terminal type dependent.

3.4.148 PATH REPLACEMENT (SEE ROUTE OPTIMIZATION)

Network Feature	
Definition	Path replacement (for ISDN QSIG), see Route Optimization (the DPNSS name).

3.4.149 PERIPHERAL UNITS ON EXTENSION POSITIONS

System Feature	
Definition	The ability to connect peripheral units to extension positions.
Settings	The CoS is set by the administrator.
Procedure and function	The equipment, for example, dictation equipment, answering machines and modems, is reached as a normal extension. Transmission of digits to the peripheral unit is possible if the caller has a keyset phone and the equipment is capable of detecting DTMF signals. The peripheral unit is called by a ringing signal and, when the calling party clears, disconnection takes place by means of a time break of the current feed to the peripheral unit.

3.4.150 PERSONAL NUMBER

User Feature	**See also the description for *PERSONAL NUMBER*.**
Definition	A Personal Number can be programmed for any extension. A Personal Number provides a user (voice extension) with different possible answering positions for incoming calls. The Personal Number service has the following characteristics: Every Personal Number can have up to 5 different lists available, (for example, at the office, at home), but only one of them can be active. Each list can be set up with up to 10 different answering positions that are selected in sequence. • When the Personal Number service is activated, incoming calls to the Personal Number are deflected to the positions in the active list, until the call is answered or is stopped for any reason. It is possible to skip answering positions depending on the origin of the call. • If the Personal Number service is deactivated, all incoming calls to the Personal Number are distributed to the assigned terminal as in a normal call without any diversion.
Settings	Personal numbers and list entries are programmed by the administrator, or by the end user via management applications.
Procedure and function	The user can activate or deactivate the service or change the active list by using a procedure. By using the Office Web application, the user can also add a new profile or adjust an existing profile. See also the boss-secretary feature where a PEN function key in a monitoring DTS or IP phone is used to change the active call distribution list.

3.4.151 PRIORITY DISCONNECT (BASED ON FRL/TCM)

System Feature **Network Feature**	
Definition	Priority Disconnect is a function that can be used for “control telephony” calls, which have been assigned a Call Service Information (CSI) class (Emergency, Priority or Routine class), in order to ensure that trunk congestion does not prevent essential “control calls” from succeeding. The function is automatic, if prerequisites are fulfilled.
Settings	The Priority Disconnect function must be activated by the administrator per system (AS parameter). FRL/TCM CoS must also be set, per user and/or route, or dedicated common abbreviated numbers, to give the calls a CSI class, i.e. a priority.
Procedure and function	The Priority Disconnect function must be activated by I/O administrator. An FRL (and thus a CSI) can be associated as a CoS to each extension and route, and also via specific common abbreviated numbers dialed by the caller, and is sent as a TCM in the private network. There are four call types; Emergency, Priority, Routine and Normal/Administrator (which has no CSI) call. To automatically disconnect a trunk call, the calling user must have an FRL/TCM (and CSI) value representing Emergency or Priority, and the disconnected call must have a lower or equal CSI value. Control calls with CSI class Emergency cannot be disconnected. Priority can disconnect another Priority call. Routine call can be disconnected by both Emergency and Priority. The disconnected call gets no warning or indication that the call will be released. The function is supported for ISDN, DPNSS, H.323, SIP and certain MFC (TL22, TL37) tie-lines.

3.4.152 PRIORITY ROUTING

Network Feature	
Definition	It is possible to classify circuits within a route, or a whole route, with respect to permitted access from an extension or an incoming external line. A specific CoS, routing access, is associated with extensions, external lines, and operators. This information is used when selecting an outgoing circuit in the own system and is sent forward with the call request.
Settings	Priority routing is set by the administrator.
Procedure and function	-

3.4.153 PRIVATE NETWORK ROUTING

Network Feature	**See also the description for *NETWORKING.***
Definition	The feature provides high capacity routing and number conversion capabilities for a private network. A number of the network features are proprietary to MX-ONE and ASB 501 04.

Network Feature	See also the description for *NETWORKING.*
Settings	Private network destinations and their required individual number translations can be set by the administrator.
Procedure and function	Works as a preprocessor to the basic routing software. Alternative routing is performed based on the dialed number, which is compared with entries in the private network routing database. If the route choice is specified to use individual translation, the translation number is fetched from the private network routing database.

3.4.154 QUEUE POSITIONS FOR OPERATORS

Operator Feature	
Definition	The ability for a call to be placed in a queue. The operator(s) can place calls in a que, and see the number of calls queued to common operators.
Settings	The number calls that are accepted in queue can be set by the administrator.
Procedure and function	The total number of queuing calls is shown on all attendant console/clients, either the whole queue, individual queue or the trunk queue. All operators also have an individual queue that shows recalls and calls to the operator's individual number. The trunk queue contains the calls arriving through the public network.

3.4.155 RECALL TO OPERATOR

Operator Feature	For more information, see the operational directions for *PBX OPERATOR TRAFFIC.*
Definition	Extended calls not answered within a certain time go back to the operator.
Settings	The time before a call is recalled to operator can be changed by the administrator.
Procedure and function	A call extended by the operator, which has been camped on for 60 seconds or not answered within 30 seconds, recalls the operator. A recall is indicated with a specific indication. Parked calls also lead to operator recall.

3.4.156 RECORDED VOICE ANNOUNCEMENT

System Feature	See also the description for *RECORDED VOICE ANNOUNCEMENT*.
Definition	The *Recorded Voice Announcement (RVA)* feature allows announcements to be supplied to incoming calls to an operator (operator announcement). It also allows announcements to be supplied to a calling party for the following call cases: • Called party has activated Follow-me • Called party has activated Direct Diversion and Diversion on Busy to an extension • Called party has activated external Follow-me • Called party is a Group Hunting Group • The call is placed in a Group Hunting Group or operator queue for a specified amount of time. • Vocal guidance for analog and digital extensions when the caller encounters certain traffic cases. When there is no available RVA resource in an announcement group, the call is still processed, but no voice announcement is supplied to the calling party.
Settings	There are two implementations and configurations, either with stored wav-files (with MGU or VSU HW), or with streaming via SIP (using Media Server). The allocation of the recorded voice messages or streaming sources is set by the administrator. Settings can be separate for trunk calls and internal calls.
Procedure and function	If initiated, the recorded voice announcement can be played back automatically, for some selected traffic cases.

3.4.157 REDUNDANCY, SERVER AND NETWORK

User Feature	For more information, see the description for *NETWORK REDUNDANCY and SERVER REDUNDANCY*.
Definition	MX-ONE provides high availability by offering Network Redundancy and Server Redundancy. **Network redundancy** One type of network redundancy are supported with MX-ONE: • Ethernet bonded network redundancy Ethernet bonded network redundancy uses a switched network for network redundancy. **Server redundancy** A standby server can take over the tasks of an ordinary server suffering from, for example, hardware failure. This way, a faulty server can be replaced with a minimum of disturbance. When using server redundancy, ordinary servers and an additional, standby server are grouped as a cluster. The standby server is prepared with data from the ordinary servers in the cluster and ready to start an instance of any of these servers in case of a server fault. See also HLR backup/redundancy.
Settings	Redundancy options are set by the administrator.
Procedure and function	

3.4.158 REFER BACK/TOGGLE

User Feature	**See also the description for *THREE-PARTY SERVICES*.**
Definition	The ability of an extension in inquiry mode to refer back and forth between the inquiry and the original call. The connected/parked party can be an internal party, a party in the private network or an external party.
Settings	-
Procedure and function	Every time the extension wants to alternate an inquiry call, the refer back procedure is used.

3.4.159 RELEASE PRINCIPLES

System Feature	
Definition	**First party release:** An established connection is released when either the calling or the called party terminates the call. **Calling party release:** An established connection is released when the calling party terminates the call.
Settings	Route setting by the administrator.
Procedure and function	**First party release** When any party goes on hook (or equivalent action) the connection is automatically released. The remaining party is given busy tone. **Calling party release** When the calling party goes on hook first the connection is released automatically. The called party, if still remaining, is given busy tone. If the called party goes on hook first the connection remains. Time supervision is started in the calling party's exchange. If the called party goes off hook before the time supervision expires, speech connection is again established. If time supervision expires, the connection is released.

3.4.160 REMOTE EXTENSION (MOBILE EXTENSION)

User Feature	**See also the description for *REMOTE EXTENSION*.**
Definition	The *remote extension (also called mobile extension)* feature makes it possible to have public terminals as extensions in MX-ONE. Any public terminal which calls a predefined number in the PBX by dialing either manually, using the phone book, or the calling card service will after validation receive a dial tone from the PBX. The validation can be made either by using the caller's A-number or by entering the caller's PIN code (authorization code). The public terminal will then get full access to the functionality and features as for a generic extension. Also internal parties which make calls to the remote/mobile extension retain full functionality, such as call back, or camp on from operator. It is possible to define additional public terminals which can alternate as answering position by dialing a procedure.

User Feature	**See also the description for *REMOTE EXTENSION*.**
Settings	Remote extensions are initiated by the administrator.
Procedure and function	-

3.4.161 REPEATED DEFLECTION

System Feature	
Definition	The purpose of the feature *repeated deflection* is to provide the exchange with a platform for the *repeated individual diversion* and the *personal number features*. The user does not need to initiate anything, because the feature repeated deflection does not provide any service by itself.
Settings	-
Procedure and function	Both Repeated Individual Diversion and Personal Number can only be defined on individual directory numbers. That directory number is called a Personal Number. Each Personal Number can have one or more lists. Each list contains answering positions. When that directory number is called and any of those services is active, the call is deflected to the answering positions until any of them answers the call, or the call is stopped for any other reason. If none of the two services is active, the calls are distributed to the assigned terminal as for a normal call.

3.4.162 REPEATED INDIVIDUAL DIVERSION (RID)

User Feature	
Definition	The aim of the RID is to provide users, that is voice extensions, with different possible answering positions for the incoming calls. Each individual extension directory number having a terminal assigned, can have RID applied. This feature is standard for cordless and IP terminals. The RID feature has the same characteristics as the Personal Number service, except that it can only have one list.
Settings	See the *Personal Number* feature.
Procedure and function	See the *Personal Number* feature.

3.4.163 REROUTING

System Feature **Network Feature**	
Definition	When a call from an external line encounters congestion, a vacant Number, a busy extension, an internal failure, an extension that is not available, or an extension that does not reply, it is possible to program routes or individual external lines for rerouting to an answering position.
Settings	Rerouting options and destinations are set by the administrator.
Procedure and function	Depending on the incoming route category, when a call encounters congestion, a vacant Number, a busy extension, an internal failure, an extension that is not available, or an extension that does not reply, a decision is made as to whether the call is to be rerouted. Different rerouting numbers can be set for a day or night switched system. Different rerouting numbers can be set per customer in the customer group. The feature can be networked.

3.4.164 RING SIGNALS

System Feature	
Definition	The ability for a user to receive different ring signals (on called extensions) depending on the type of call: • Internal, ring signal used when receiving internal calls. • External, ring signal used when receiving external calls. • Callback, ring signal used when the callback function is enabled. • Diverted calls, ring signal used when the call has been forwarded to an answering position. • Distinctive ringing, ring signal used when a SIP extension has an additional phone number on the same line as an existing number. • Continuous ringing, ring signal used for calls from attendant (only in specific markets).
Settings	Market setting which may change ring signals can be done by the administrator.
Procedure and function	-

3.4.165 RING SIGNAL PER CALLING NUMBER OR ROUTE NUMBER

User Feature	
Definition	The purpose of the basic function ring signal is to provide a specific ring tone (internal, external, or callback) to the called party, depending upon the calling party or incoming route number. The callback ring signal, which would normally only be used at recall to a requester of the Callback service, can be used as an 'alarm ringing' indication to the called party, if such a function is defined in the system.

User Feature	
Settings	Ring signal can be set by the administrator. An 'Alarm ringing' configuration can be set by the administration, if appropriate routes and external applications are defined.
Procedure and function	Ring signals can only be initiated on directory numbers (for extensions). A ring signal can be initiated for a called directory number when the (A-party) caller is either internal, set by the caller number, or external, set by the route number. The ring signals are terminal type dependent.

3.4.166 ROUTE OPTIMIZATION (ISDN/DPNSS)

Network Feature	
Definition	Route optimization (for ISDN called: Path replacement) between exchanges can be executed when a direct route with free lines exists between two exchanges, but when the original call is set up via another route passing a third exchange (or more exchanges).
Settings	The Route Optimization is set by the administrator.
Procedure and function	Events that can lead to route optimization are: • Extending/transfer has occurred in a third exchange • Alternative routing • Conference terminated with two external parties remaining Route optimization between the exchanges means, for example, that a call from exchange A that is answered by exchange B and thereafter extended to exchange C shall utilize a direct route, that is the connection shall be set up directly between exchanges A and C. The route to exchange B is cleared down. Note: IP tie-lines do not support route optimization.

3.4.167 ROUTING, ALTERNATIVE ROUTING

Network Feature	
Definition	The ability to reach external destinations using different routes.
Settings	Alternative routing is set by the administrator.
Procedure and function	The system uses sequential hunting on the ordinary route and the alternative routes; that is, when the ordinary is fully occupied, the system starts hunting in the first alternative route, and so on. The system can add and discriminate programmed pre-digits; that is, if a route to another PBX is fully occupied and the call has to be switched through the PSTN, the system adds the extra digits needed. A maximum of 20 pre-digits can be added to each alternative route, and the total code may not consist of more than 34 digits.

3.4.168 ROUTING, OVERFLOW

Network Feature	
Definition	The process of routing a call automatically through another (second choice) route from the system, when a call cannot find a free circuit in a first choice route. There may also be an overflow, in the same system, from a second choice route to a third choice, and so on.
Settings	Set by the administrator.
Procedure and function	If there are no free circuits in the requested route, overflow can occur to alternative routes. In the originating system, overflow can occur to a maximum number of seven alternative routes; that is, the originating system permits eight choices to the requested destination.

3.4.169 ROUTING, REPEAT ATTEMPT

Network Feature	
Definition	A second attempt to set up a connection for a call, from where the first attempt encountered difficulty when setting up the connection.
Settings	Routing is set by the administrator.
Procedure and function	A repeat attempt can be made if, for example, the seize acknowledgement signal is received either too early or not at all. If the acknowledgement signal is not received at all, it is interpreted either as a line fault or that the seized exchange is unable to respond to the seize signal.

3.4.170 ROUTING, RETURN BLOCK

Network Feature	
Definition	An exchange is not permitted to route the call to another circuit in the route from which the call originated.
Settings	Routing is set by the administrator.
Procedure and function	-

3.4.171 ROUTING SERVER

System Feature	
Definition	In a private network where IP trunks are used, the routing server feature significantly reduces the effort to program the data needed for optimized routing of calls. Management of routing data, e.g. IP addresses and alternative routing data, for the whole network is only done in the Routing server. This routing data is automatically downloaded to all routing server clients (other MX-ONEs in the network).
Settings	The Routing Server is set up by the administrator.
Procedure and function	The *routing server* feature has two parts, the server and the satellite (or client). The server stores the IP routing and alternative routing information on a permanent basis (reload data) whilst the satellites store the information on a temporary basis which is dynamically updated to reflect the changes and network status in the server.

3.4.172 SECURITY

System Feature	
Definition	The security measures available for the MX-ONE system are mainly based on the following open standard technologies: • **SSL (or TLS)** - The Secure Socket Layer (SSL) or Transport Layer Security (TLS) provides secure access to IP phones and web services and secure signaling between IP phones and MX-ONE Service Node (SN). • **SSH** - Secure Shell (SSH) provides secure console-based access to IP phones and the SN. • **IPSec** - IP Security (IPSec) is used to protect the signaling messages exchanged between Servers (SN). • **SRTP** - Secure Real-time Transport Protocol (SRTP) is used to protect the media streams of the voice communication.
Settings	This feature is set by the administrator.
Procedure and function	Certain IP terminal models support encryption of both signaling and media, and can indicate if a call uses SRTP, i.e. encryption of voice media, for example by showing a padlock icon. For gateway calls known as a gateway connection by the system where the terminal is registered, the padlock icon is not shown, because the other end-party (or some part of the connection) can have an non-encrypted connection leg.

3.4.173 SERIAL CALL

Operator Feature	
Definition	The ability of an attendant/PBX operator to regain an external call when the called party terminates the call.

Operator Feature	
Settings	-
Procedure and function	Incoming and outgoing external calls can be serial call marked by the operator. When the not serial call marked party terminates the call, the operator is recalled. If the serial call marked party terminates the call the operator is not recalled or informed.

3.4.174 SHORT MESSAGE SERVICE (SMS)

User Feature	
Definition	The Short Message Service (SMS) performs transport of text messages (maximum 160 characters). These text messages can be received in any call state, that is, an SMS message can be received during an ongoing call. An SMS server is a software application running on a PC with Ethernet connection to the Network Interface Unit. There can be one SMS server per Server (LIM). A proprietary software driver is needed for communication between the software application and the TCP/IP-interface on the PC.
Settings	SMS permission is set by the administrator. Diversion and Personal number options for SMS are set by administrator.
Procedure and function	In the ASP113 the primary users of the SMS feature are Cordless DECT phones, remote extensions, soft-phones and attendants.

3.4.175 SIMPLIFIED INTERCEPTION

Operator Feature User Feature	
Definition	The ability of an operator or secretary to make phone tending more efficient, which means that the operator or secretary can give callers more informative messages. Simplified interception service requires no extra equipment.
Settings	
Procedure and function	The attendant console/client, or the secretary's' phone display the absence information. The procedures do not include handling of message types other than absence information, and there is no feature for displaying the names of users. Users with digital phones with display can also see the absence information when calling an absent user extension.

3.4.176 SINGLE NUMBER INDICATION (SNI)

User Feature	
Definition	The ability to set an extension number that is displayed on the other party's display. This means that the B-extension always receives the same directory number independent of which phone the A-extension is using, if all the phones of A has the same additional number. This also applies for calls in the opposite direction. To offer this possibility, each extension should have both a directory number and an additional number.
Settings	This feature is set by the administrator.
Procedure and function	-

3.4.177 SILENT RINGING

User Feature	
Definition	The ability of a user to activate a *silent ringing* feature by pressing the Mute key on a digital phone or IP phone. A flashing LED will indicate any incoming calls to the phone.
Settings	-
Procedure and function	If the Mute key is pressed during idle, the tone ringer will not be activated for the subsequent call(s). The *silent ringing* feature will be active until a feature in the phone deactivates it, for example, the user lifts the handset or presses the Mute key. The procedure is terminal type dependent.

3.4.178 SNMP, SIMPLE NETWORK MANAGEMENT PROTOCOL

System Feature	**For more information, see the Operational Directions for *MIVOICE MX-ONE SNMP SUPPORT, ALARM NOTIFICATION AND EMERGENCY CALL EVENTS.***
Definition	The SNMP (Simple Network Management Protocol) is an Internet-standard protocol used to manage devices on IP networks. As defined in the RFC 1157, Network management stations execute management applications which monitor and control network elements.
Settings	This feature is configured by the administrator.
Procedure and function	The MX-ONE Service Node (SN) can be supervised from a Simple Network Management Protocol management system and configured to send alarm notifications by SNMP traps or optionally by e-mail or text messaging using Short Messaging Service (SMS). There are three SNMP MIB implementations available in the SN; the old deprecated general Ericsson MIB, the newer general Mitel/Aastra MIB, and a Mitel Emergency Call MIB. The SN can be monitored from an SNMP management system, using one or two or all of them. The SNMP implementation in the SN is based on net-snmp.

3.4.179 SOFT-KEYS

User Feature	
Definition	Advanced pre-programmed keys for the most used features and programmable keys for other features. The functionality of the soft keys is to indicate for the user which features are available from the phone at a certain traffic case (for example, idle, busy, calling).
Settings	Soft-keys are set by the administrator
Procedure and function	The user can request any of the available features by pressing the corresponding soft key. The procedure is terminal type dependent.

3.4.180 SPEED DIALING

User Feature	
Definition	Speed dialing is the ability for users to make calls by dialing a speed dialing number which is automatically translated to a full number and used as destination number. Speed dialing can be common to the system for all users, or individual for each user (only on non-generic extensions).
Settings	Common speed dialing numbers are set by the administrator. Individual speed dialing numbers are set by each user.
Procedure and function	A speed dialing number is a 2–10 digit number. The numbers are divided into four tables; extensions can be allowed to use some or all tables. Common speed dialing numbers can be made semi-automatic by programming incomplete numbers, where the extension can add digits to complete the number.

3.4.181 STATIC SEMI-PERMANENT CONNECTION (SSPC)

System Feature	
Definition	The SSPC feature allows the establishment of a nailed up connection between two MX-ONE individuals in the system.
Settings	Set by the administrator.
Procedure and function	Static semipermanent connections are initiated by command. The command contains the two equipment positions to be connected. The two positions will be connected ignoring any restrictions such as CoS or traffic matrix. The feature supports digital trunks, CAS extensions and ISDN S0 extensions.

3.4.182 STREAMING ON IDLE EXTENSION

System & User Feature	
Definition	The Streaming/Music on idle extension (MOI) feature facilitates streaming of voice media to SIP extensions which are in idle state, i.e. have no ongoing call, or other activity which makes the terminal busy in some way. The MOI function can be used for example for Music in an idle extension, via the loudspeaker (or handset/headset).
Settings	Configured by the administrator. Requires both system settings and individual extension key settings.
Procedure and function	Streaming /Music on idle extension is a function which is initiated by command or via SNM&PM. The feature requires both configuration of a Media Server (soft MGW) which shall act as a “streaming server” collecting the media streaming sources, and configuration of a specific key on Mitel 6900/6800 SIP terminals. The key provides a menu function in the phone display, where the end user can select which source to use, or turn off the function. wav, ogg and mp3 formats are supported.

3.4.183 SUB-ADDRESSING

User Feature	
Definition	The ability for an ISDN S0 extension or ISDN trunk to send and receive extra address information in the form of a sub-address related to the called/calling/connected party number. This information is transferred transparently through MX-ONE.
Settings	-
Procedure and function	-

3.4.184 SURVEILLANCE, OBSERVATION AND MONITORING (SOM)

System Feature	**See also the description for SURVEILLANCE OBSERVATION AND MONITORING.**
Definition	The Surveillance, Observation and Monitoring feature comprises a number of functions to observe and monitor traffic to and from certain objects for surveillance purposes. These objects can be individual extensions, group numbers, routes (trunk groups), or destinations in a private or public network. All traffic associated with observed objects is intercepted and data collected in the form of traffic events. The SOM also includes functions to monitor the observed object’s speech channels.
Settings	The functionality is configured and initiated by the administrator. It requires dedicated trunk resources and appropriate external Surveillance Control Center applications for the SOM functions.

System Feature	See also the description for SURVEILLANCE OBSERVATION AND MONITORING.
Procedure and function	The functions are controlled by external third party application(s) via service requests and events, using a specific SOM protocol.

3.4.185 TELEPHONE NAME SELECTION (TNS)

User Feature	
Definition	The ability to connect any function key with a phone number and label it with a name. The function key is named TNS key. For more information, see also the feature *Dial by Function Key* in this document.
Settings	Set by the administrator.
Procedure and function	When the user presses the function key, the phone number is automatically dialed.

3.4.186 TERMINAL PORTABILITY

User Feature	
Definition	The ability to move an ISDN S0 terminal from one wall socket to another within the same basic access, when the call is active. It also allows the user to move a call from one terminal to another terminal within the same access, when the call is active.
Settings	-
Procedure and function	-

3.4.187 THROUGH-CONNECTION AT POWER FAILURE AND PROCESSOR BREAKDOWN

System Feature	
Definition	Automatic connection of analog external lines to predefined analog extensions in the event of power failure or processor malfunctions. The extension and the external line must each be connected to the power failure board (required FTU) with separate cables.
Settings	Which analog extensions and trunks that shall be connected at power failure is configured by administrator.
Procedure and function	If an extension has an active call when the fault arises, the speech connection will be broken and the extension is connected directly to the specified external line.

3.4.188 TRAFFIC CONNECTION MATRIX

System Feature	
Definition	A check is made in the system to verify that the interconnection of two parties is permitted. The parties may be extensions, hunting groups, conference equipment, or external lines. The check is done in a 16x16 matrix.
Settings	The matrix is programmed by the administrator.
Procedure and function	-

3.4.189 TRAFFIC RECORDING

System Feature	**See also the description for *TRAFFIC RECORDING*.**
Definition	Traffic recording of a number of different measurement objects, for example, voice extensions or routes. The maximum number of ongoing measurements in parallel is 250. For each measurement, a measurement result is obtained that is stored in the program memory and also on a storage unit. The collection of measurement data from the measurement objects is initiated automatically at each quarter of an hour. A congestion monitoring value can be initiated/altered for the objects route group.
Settings	Traffic recording is initiated by the administrator. Each user must also be allocated a traffic connection CoS.
Procedure and function	-

3.4.190 TRANSFER/CALL TRANSFER

User Feature **Network Feature**	**See also the description for *THREE-PARTY SERVICES*.**
Definition	The ability of an extension to transfer a call to another party in the own system or in another connected system, or to an operator or external line. The networked feature *call transfer* is only available within a private network with appropriate protocol. It is possible to transfer both before an answer, or during an active call in speech. In both cases there must be another call on hold.
Settings	Permission for *transfer to trunk* and *transfer before answer* is set by the administrator. Prohibition to transfer outbound public trunk calls can be set by the administrator.
Procedure and function	The extension first initiates an inquiry call or equivalent parking procedure, and when the new call is established or alerting the transfer procedure, which differs for different types of parties, is used.

3.4.191 TRANSIT COUNTER (LOOP AVOIDANCE)

Network Feature	
Definition	A transit counter is used in ISDN/H.323 networks (called Loop Avoidance counter in DPNSS) to prevent call being routed in loops.
Settings	The number of allowed transit exchanges can be set by the administrator.
Procedure and function	For each route a value can be set that defines the maximum number of transit exchanges a call can be routed through, to reach its final destination.

3.4.192 TRANSIT TRAFFIC

Network Feature	
Definition	The ability to transit switching in the exchange, i.e. to handle trunk-to-trunk calls.
Settings	The route programming is set by the administrator.
Procedure and function	Incoming external line calls can be switched to other external lines, either automatically or by the operator.

3.4.193 TRANSMISSION MATRIX

System Feature	
Definition	A matrix that defines the level of digital amplification/attenuation to be used for a connection between different types of (legacy) interfaces, in order to adjust the speech/media to a suitable level.
Settings	The matrix and the transmission CoS per extension/route is set by the administrator.
Procedure and function	The transmission matrix is only relevant for non generic extensions and TDM trunks.

3.4.194 TRUNK CALL DISCRIMINATION (TCD)

System Feature	
Definition	The ability to block extensions from dialing specific numbers. The feature is primarily used to block calls to certain public network destinations, e.g. international calls, but can be used generally for any call.
Settings	Each extension is given a Class of Service with one of 15 possible categories for day traffic and 15 possible categories for night traffic. Information about which numbers a user with a specific CoS is permitted to dial is set up by the administrator.

System Feature	
Procedure and function	The control applies to any digits dialed by an extension, e.g. both external and internal calls can be barred. Extensions can be assigned separate categories for day and night switched systems. The check can be made for up to ten dialed digits. If an extension attempts to call a forbidden number a tone message is received.

3.4.195 VIRTUALIZATION

System Feature	**See also the description for *MIVOICE MX-ONE SERVICE NODE VIRTUALIZATION*.**
Definition	The MX-ONE Service Node (SN) software can execute in a virtualized environment, where the physical server hardware can be shared between several SN call managers. In a virtualized environment, an abstraction layer, called a hypervisor, is installed between the physical hardware (host server/machine) and the operating system (guest machine). This abstraction layer allows that more than one guest machine co-exist on the same physical hardware, in order to share resources like memory, CPU, etc. These guest machines are commonly referred to as Virtual Machines (VM). VMWare software is used as part of the virtualized solution. (KVM, which is included in Linux, is a possible alternative). Mitel has three VMware software options to run MX-ONE Service Node in a virtualized infrastructure: - Consolidated setup - Availability setups with VMware vSphere Availability - VMware vSphere High Availability - VMware vSphere Fault Tolerance
Settings	The virtualization configuration is set up by the administrator.
Procedure and function	-

3.4.196 VIRTUAL GENERIC EXTENSION

System Feature	
Definition	A generic extension which is not associated to any terminal type. A virtual generic extension is created when a directory number with a common service profile is initiated in the exchange.
Settings	-
Procedure and function	A virtual generic extension can be used by temporary users such as consultants who normally do not have a wired extension. The number of virtual generic extensions is related to the number of directory numbers per Server (LIM) and not to the physical number of terminals or positions. After the basic setup has been initiated for the directory number, different applications can be defined to affiliate this directory number to a terminal.

3.4.197 VOIP QUALITY OF SERVICE LOGGING

System Feature	
Definition	The Voice-over-IP Quality of Service (QoS) monitoring feature collects data concerning end-to-end delay, jitter, and packet loss for RTP media traffic. For most IP end points (H.323 terminals and trunk lines, Attendant OMDs, and Media Gateways), the collection is done per call. The collected data is stored in a comma separated text file using the Call Information Logging output feature. The QoS feature can be switched on or off using I/O commands. Limitations: SIP end points, H.323 soft clients, and some older H.323 terminals cannot provide QoS data.
Settings	QoS is set by the administrator.
Procedure and function	

3.4.198 VOIP RECORDING

System Feature **User Feature**	**For more information, see the description for *VOICE RECORDING*.**
Definition	The Voice-over-IP Recording services provide active VoIP recording for the MiVoice 4xxx, Mitel 7xxx terminals and Mitel 6900/6800/6700 terminals.
Settings	VoIP recording of unencrypted or encrypted calls is set up by the administrator by configuring the telephone config-files. For recording of encrypted calls, a parameter setting of the CSTA interface is required. A record-on-demand key can be configured on the IP telephones (certain models). Policies for recording are the responsibility of the recording system.
Procedure and function	The recording is controlled via external application(s) for VoIP recording, like the MiVoice Call Recorder application, or similar third party applications.