

MiVoice MX-ONE System Planning

DESCRIPTION



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SUMMARY

This document provides an overview of the planning activities that have to be carried out before installing MX-ONE and integrating it with the IS/IT infrastructure.

Some of these activities are common to the whole solution and are therefore described in detail in this document, while other activities are specific to one of the MX-ONE components and therefore only described briefly in this document, with a reference to the detailed description.

Some of the planning activities are not relevant or only partly relevant for the MX-ONE Private Cloud solution, for example the sections on certain legacy devices that are not relevant in a Cloud solution, but the document still covers both the CPE and Cloud solutions.

If virtualization is used, also see VMware documents regarding planning matters.

2

INTRODUCTION

The MX-ONE is made up of a number of components that have to be combined and integrated with an existing IS/IT infrastructure.

For this purpose, several aspects have to be taken into account and several activities need to be carried out, for example:

- Planning and configuration of the corporate Local Area Network (LAN)
- Connection to public telephony networks, such as the Public Switched Telephony Network (PSTN) and the Public Land Mobile Network (PLMN).
- Integration with existing fault and performance management systems, business automation systems and all collaborative systems/services, e.g. MS Exchange collaboration services.
- Planning of redundancy functions to be used.
- Planning of Security.
- Planning the use of IPv6 vs IPv4.

2.1

SCOPE

This document is intended for network planners and administrators, and is used when integrating the MX-ONE in the enterprise IP network.

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

2.2

GLOSSARY

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

2.3

SYSTEM OVERVIEW

The MX-ONE comprises several components that provide different functionality and services. The components can be combined according to the specific customer needs. There are basically two commercial offerings; one CPE offering, where the equipment is located at the customer premises, and one for a Private Cloud offering, where the equipment is located at a service provider, and “leased” to the end customers.

For an overview of the MX-ONE architecture, see Figure 1.

For a list of RFCs and ITU-T standards, see the description for *MIVoice MX-ONE*.

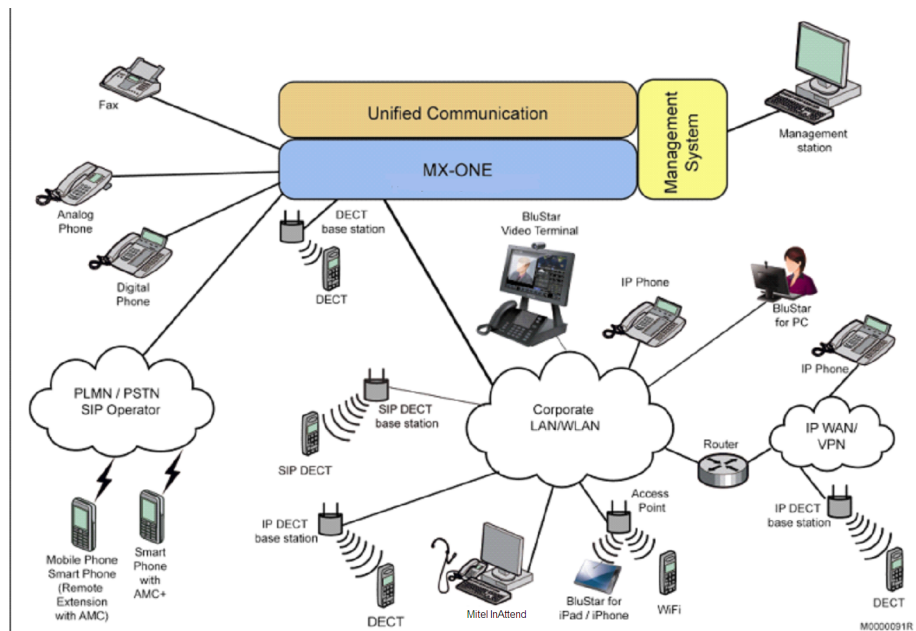


Figure 1: MX-ONE Architectural Overview

The MX-ONE is the core component of MX-ONE and provides IP telephony services, as well as traditional (TDM-based) Private Branch Exchange (PBX) functionality, including DECT functions.

The main products in Unified Communication are MiCollab Advanced Messaging, MiContact Center Enterprise and the Mitel Collaboration Management, CMG.

MiCollab Advanced Messaging delivers voice mail and fax functions. It supports Unified Messaging and cooperates with major email systems, such as, MS Exchange, Lotus Notes, and Groupwise. MiCollab Advanced Messaging is optional in MX-ONE.

MiContact Center Enterprise is an all-in-one, adaptive and flexible platform for UCC, mobility, contact center, Business Process Automation, analytics and reporting, as well as service and database integration. It offers skills-based routing functions, agent desktop applications and management applications for server-based contact centers.

The Mitel Collaboration Management, CMG consists of a number of applications that enhances the MX-ONE. The Mitel CMG includes all necessary applications for contact management, including the Mitel InAttend, attendant services, and Visit, visitor registration and management.

The MiCollab AVW Conferencing terminals and clients, which are part of Mitel's UCC MiCollab portfolio, are an integral part of the solution.

The Mitel Mobile Client offers a contextual graphical user interface for easy access to the corporate PBX services directly from the mobile phone. When combined with the mobile extension functionality of the MX-ONE, it enables in-call PBX services (inquiry, transfer, conference, etc...) as well as absence indication, phone book lookup, voice mail access, call forwarding, and so on. Additionally, Mitel Mobile Client's Dynamic LCR and the Traveling SIM features initiate call routing based on the user's location significantly reduces mobile roaming charges for users abroad.

MX-ONE Provisioning Manager is the user and extension management application in MX-ONE, providing a single point of entry for managing user and extension data in MX-ONE, MiCollab Advanced Messaging, Mitel CMG, and Mitel Mobile Client Management System. MX-ONE Provisioning Manager is the recommended user management solution for customers migrating from the Mitel MX-ONE Telephony Switch (Mitel TSW) to MX-ONE 6.x.

Different MX-ONE Managers are used to manage the MX-ONE solution. These applications provide administrators with ease of use tools and integrates well with the existing IT environment.

For detailed information about the MX-ONE solution, see *MIVOICE MX-ONE SYSTEM OVERVIEW* and *MIVOICE MX-ONE SYSTEM DESCRIPTION*.

3 LAN AND WAN PACKET NETWORKS

There are several aspects to consider in an IP network and the following sections describe the basic topology concept briefly and also the actual planning activities that are needed. Both IPv6 and IPv4 are supported.

3.1 NETWORK TOPOLOGY

When planning to deploy the MX-ONE from a telephony point of view, it is important to understand the structure of a system and the differences between the networked nodes and the elements making up a node.

Apart from the logical structure of the MX-ONE, it is important to understand how the underlying IP networks logical structure impacts on the system solution.

3.1.1 ELEMENT STRUCTURE IN ONE MIVOICE MX-ONE

The MX-ONE can consist of a maximum of 124 Service Nodes (MX-ONE Service Nodes). Each server controls from one to 15 media gateways.

There are two types of servers and a number of media gateways. One server alternative is the ASU, Mitel ASU-II or Mitel ASU Lite server, based on an embedded server board. The other alternative is a standard server, which is a high-end, commercially available server (DELL, HP...).

For new delivery, there is one software-based, one compact, one mid-size, and one large media gateway.

It is allowed to mix the different media gateway types for the same MX-ONE Service Node.

There is an MX-ONE Media Server which emulates parts of an MGU board, and is a soft media gateway. The MX-ONE Media Server is for SIP-only scenarios.

The compact MX-ONE 1U can contain the MGU board and one other board as TMU/MFU, trunk, analog or digital extension interface board. The MX-ONE 1U can instead of above, contain the ASU server board as stand alone.

The mid-size MX-ONE Lite (3U) always contains the MGU2 board and optionally other boards. When an MX-ONE server board is mounted, board positions are also available for e.g. one MGU2, plus up to 3 TMU/MFU, trunk, analog or digital extension interface boards. The chassis have in total 6pcs of 20mm board positions.

The MX-ONE Classic is the large unit contained in a 7U high 19 inch sub-rack with space for 16 device boards for classical telecom interfaces (digital trunks, IP trunk, IP extensions, mobile extensions, digital and analog extensions).

One Server can handle up to fifteen media gateways of the MX-ONE 1U, Lite, Classic or Media Server type.

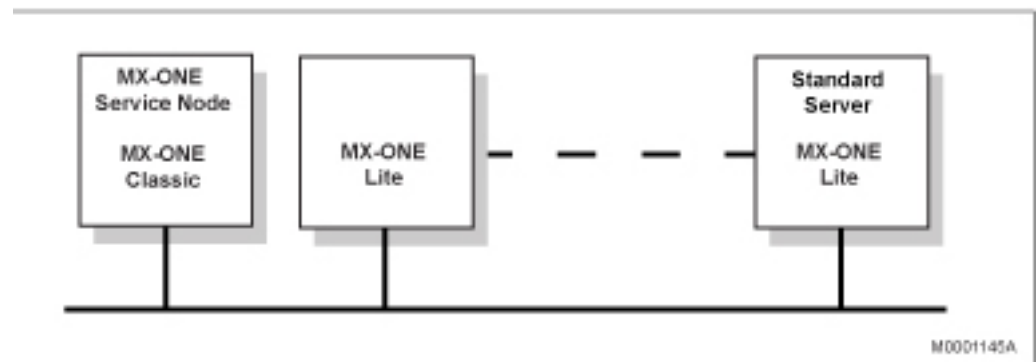


Figure 2: MX-ONE Service Nodes within one Node

Each MX-ONE can represent a node of a more complex PBX system, where the different nodes can communicate.

All MX-ONE Service Nodes within one node communicate on a peer-to-peer basis, using a proprietary signaling protocol. All MX-ONE Service Nodes within the same node are managed as one entity and are served by one entity of Mitel CMG, MX-ONE Provisioning Manager (PM), and MX-ONE Service Node Manager (SNM).

MX-ONE Service Nodes and media gateways communicate over Ethernet. The length of individual network segments can be maximum 100 meters, according to Ethernet standards.

3.1.2

NETWORKED NODES

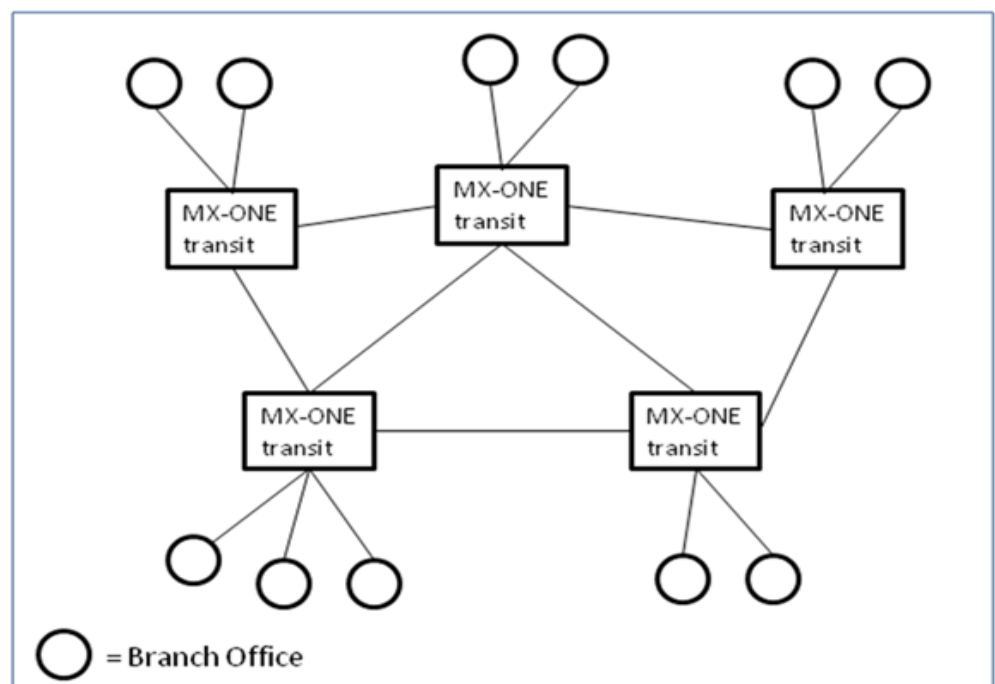


Figure 3: Networking Nodes

The figure 3 represents a traditional, TDM based (e.g. Q-SIG) large private network. Transits are used to concentrate long distance traffic and to minimize the programming effort for routing data as the Branch offices only need to know a single route for all destinations belonging to the private network.

Transits are often connected in a meshed fashion in order to allow alternative routing when back-bone links are congested or out of service.

This structure is not optimal for IP based private networks as the media will be established directly between the end point exchanges, but the signaling will always remain through the transit exchanges. For a customer that migrates from TDM to IP, it can be beneficial to keep the structure and existing programming until a larger part of the trunks has been changed to IP.

With IP, all exchanges in the network are connected directly to an IP back-bone WAN over a single route, or via two routes for redundancy purposes.

To minimize programming of routing data in the exchanges, it is recommended to use the Private Network Routing feature and the Routing Server application if H.323 tie-lines are used.

For SIP based private networks, it is recommended to use the E-Num feature, where the routing data is held in a DNS type server, which will return the IP address of the destination exchange when queried for each call.

3.1.3

IP NETWORK STRUCTURE

The current best practice to carry data from point-to-point without major disturbances is to deploy a switched architecture and to limit routing as much as possible.

Multi-media traffic of different types have different network requirements, and this must be considered when designing the network. All links must be able to carry the additional voice traffic with good margin when added to the existing data traffic. Telephony traffic is symmetrical whereas data communication is fairly asymmetrical, which can easily cause trouble in certain types of network links, that are designed for asymmetrical data traffic.

In the IP network, plan for the extra load of data traffic that is caused by the telephony elements added to the network. The structure should be built in such a way that 80% of the estimated traffic is carried inside the logical network that is supposed to carry the traffic. To realize this target, the recommendation is to implement the telephony service in a Virtual LAN (VLAN) structure. This makes it easier to control the behavior and to avoid heavy traffic collision in the routers.

To reduce the effects of broadcast traffic when deploying IP telephony using the MX-ONE™ Service Node, it is recommended that VLANs are used for IP phones.

It is strongly recommended that the number of IP phones in a subnet is limited to a maximum of 510 devices. The reason is the amount of ARP requests in certain restart situations. For some installations (for example, contact centers) the traffic density may require a lower number of IP phones per subnet.

For an example of a typical IP network with redundancy and Wide Area Networks (WAN) connections, see Figure 4 Typical Large IP Network on page 10. The basic principle is a core routing layer and independent distribution layers for clients and servers.

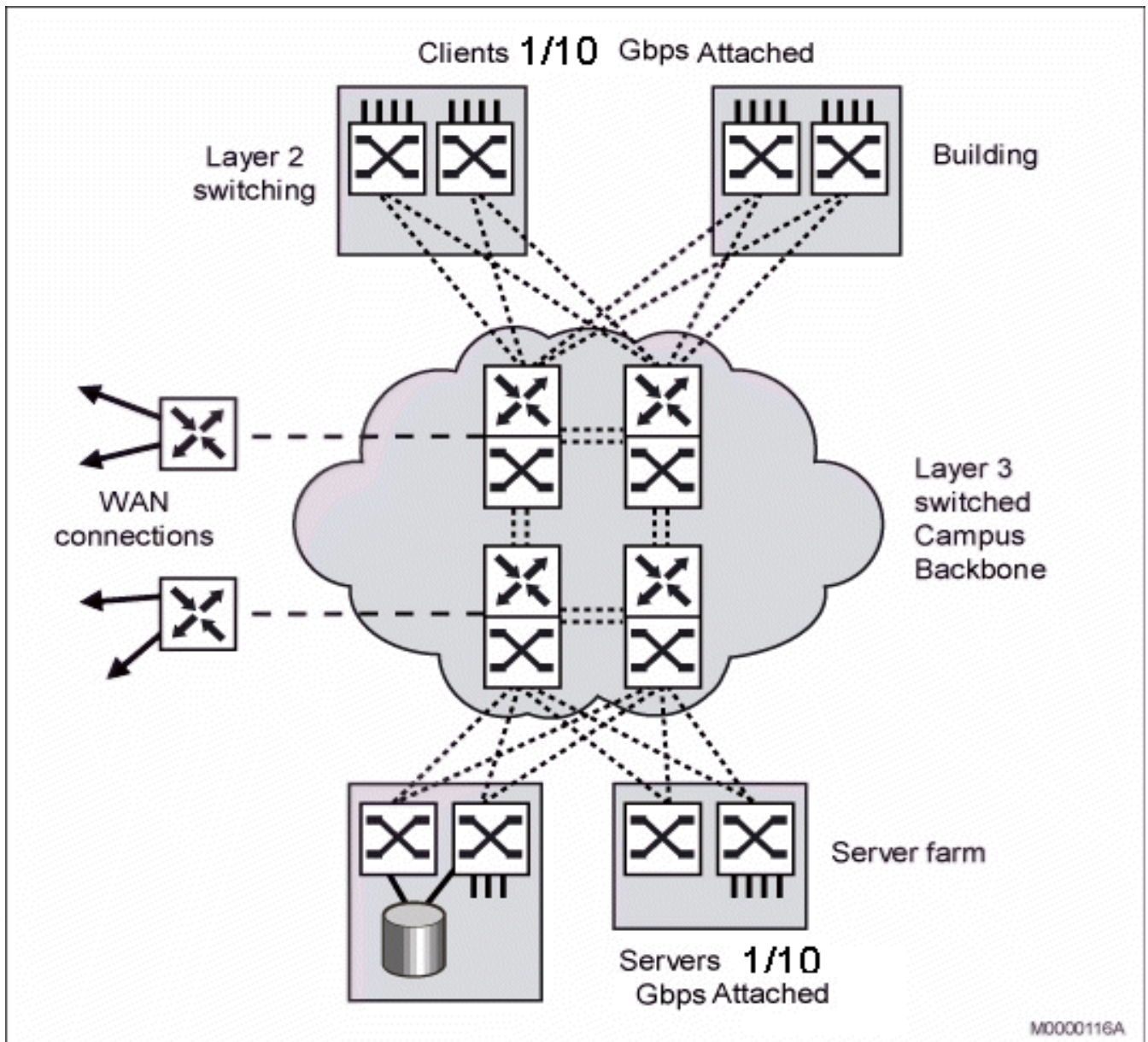


Figure 4: Typical Large IP Network

One of the most important and critical factors, when planning for an IP network to include voice and telephony traffic, is Quality of Service (QoS). For an overview of where different QoS policies are derived and deployed, See Figure 5. Not implementing structured QoS policies may result in degradation of some media services, which is especially critical where low speed WAN links are used.

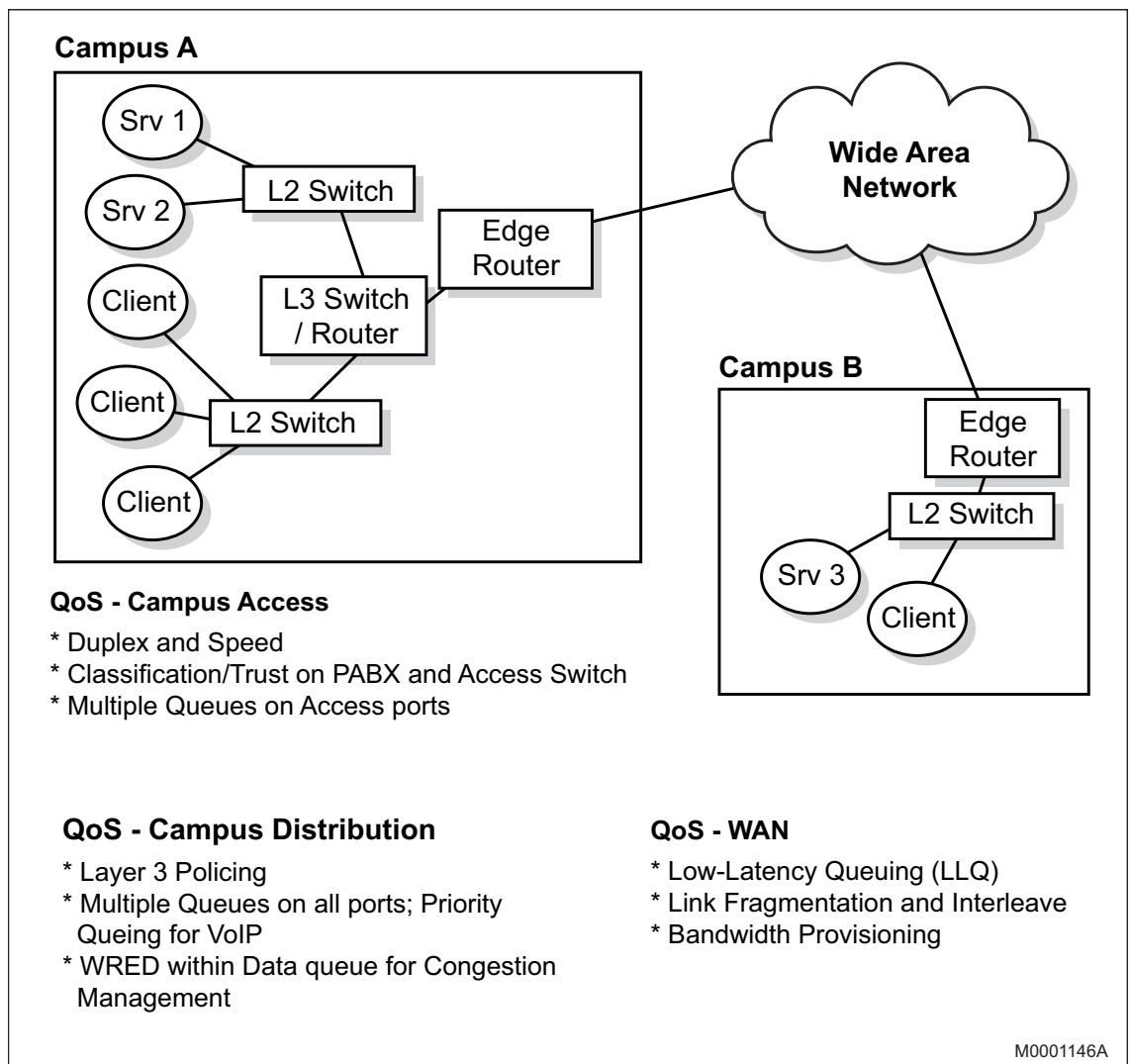


Figure 5: Needed Quality of Service Policies

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

Except for the Call Admission Control (CAC) feature, the MX-ONE is not equipped with any mechanism that limits or blocks incoming or outgoing calls to a Server - Media Gateway depending on the network conditions (temporary or local congestion) or assigned bandwidth. For this reason, it is not recommended to locate MX-ONE Service Nodes in separated LANs, if it cannot be guaranteed that the network (WAN) connecting them has the required parameters (allocated bandwidth, maximum delay, maximum jitter, maximum packet loss) that are necessary for this type of application and the maximum estimated traffic flowing through the connecting WAN.

3.1.3.1

Call Admission Control

The Call Admission Control (CAC) mechanism can be used between IP domains. A maximum bandwidth and the codec priority is set per LAN segment for all outgoing IP calls (H.323 and SIP). The lowest common bandwidth will be used for the communication between the IP domains.

For more information, see the operational directions for CALL ADMISSION CONTROL, and also the description of IP NETWORKING.

3.1.3.2

Inter-Server Modeling

The signaling between different end-points in the MX-ONE environment can be divided into three different categories; **extension**, **trunk**, and **inter-server**. This section only covers inter-Server signaling.

Inter-server signaling is the internal signaling used between MX-ONE Service Nodes for call control, management and other common services. This signaling is carried over an IP network. There is a constant background signaling that uses approximately 20 kbit/s per Server plus an additional signaling that is traffic intensity dependent. In addition to this bandwidth, it is necessary to add the IP header bandwidth. The inter-Server signaling requires a delay not higher than 100 ms. Higher delay in the network may result in slow response times, for example Attendants may have problems extending calls.

The signaling part of the inter-Server links therefore needs approximately 60 - 220 kbit/s per 1000 extension users, for information exchange depending on traffic load. 220 kbit/s equals the peak traffic of 0.2 Erlang. But in comparison to the media flow that occurs between two nodes, this bandwidth need is regarded as low (for example, an RTP stream with G.711 PCM and 30 ms frame packaging uses 80 kbit/s (Ethernet).

Apply the rule of adding one media stream extra per link to allow for signaling.

To estimate the required bandwidth between MX-ONE Service Nodes, it is necessary to make a model of the actual network and define the amount of traffic that will be applied to each Server as well as the assumed traffic pattern that will apply between the Servers.

When the planning and the traffic patterns are laid out, it is possible to apply the required bandwidths needed between different Servers and to check if the traffic from two or more Servers will pass any critical element in the underlying IP network.

When switched network redundancy is in use, both (all) networks must fulfill the requirements. Regarding network redundancy, see also 12.2.1 Network Redundancy on page 57.

3.1.3.3

Remote Branch Node Inter-Server Modelling

The modeling for a remote branch node differs somewhat from the above general inter-server modelling, since the bandwidth is more likely to be limited, so the CAC feature may be more needed. The number of extensions in the branch node is probably rather low (<1000).

Assuming that the remote branch server has 1000 extension users, and 10% of the calls go to/from the main part of the system, with a peak traffic of 0.2 Erlang and standard holding times, you get 1.7 calls/second, i.e. 0.17 calls/second average for the 10% traffic to/from the main site.

That means you need bandwidth enough for 17 simultaneous calls, but the actual needed bandwidth depends on which codecs are used. For example G.711 codecs will require 64×2 (64 kbit/s for each direction) = 128 kbit/s per call, times 17 calls, plus the control signaling, which adds 22 kbit/s, i.e. a maximum bandwidth of circa 2 Mbit/s between the branch node and main site.

Note: When low speed WAN links are used, ICMP must be enabled in the network to support fragmentation if Inter-LIM connection is limited in frame size.

3.2 IP NETWORK SERVICES

In larger networks it is impractical to configure MX-ONE to be independent of any additional services from the network and therefore the recommendation is to use at least following services:

- Dynamic Host Configuration Protocol (DHCP)
- Domain Name System (DNS)
- Network Time Protocol (NTP)

Both IPv6 and IPv4 are supported.

3.2.1 DYNAMIC HOST CONFIGURATION PROTOCOL (DHCP)

DHCP allows computers (clients) or network devices (IP phones, printers, etc) to be assigned settings from a server in a client-server model.

DHCP clients, like IP phones, request configuration data, such as IP information (IP address, subnet mask and default route) and DNS servers addresses from a DHCP server.

In MX-ONE, DHCP is used by IP phones to provide configuration data (IP address, default route, DNS) to the telephone as well as IP address to the HTTP server that holds the configuration data and the firmware files used in the terminals. DHCP can also be used in some Windows Server applications that are part of the MX-ONE solution.

Note that DHCP is not used in the MX-ONE Service Node.

For more information regarding the use of DHCP in MX-ONE, consult the specific IP Phone or Application configuration guide.

3.2.2 DOMAIN NAME SYSTEM (DNS)

Domain Name System, DNS is a system for converting host names and domain names into IP addresses on the Internet, or on local networks that use the TCP/IP protocol.

MX-ONE systems use DNS to associate host names to IP addresses in order to perform name resolution in the network.

MX-ONE Service Node has a DNS Server running locally by default. The Local DNS server is always enabled in one MX-ONE Service Node, (normally MX-ONE Service Node 1) and it shall be used to translate names internally in the MX-ONE cluster. If the MX-ONE contains more than one MX-ONE Service Node, server 1 is the DNS master and the other servers are configured as DNS slaves. The reason for using local DNS in MX-ONE is the fact that a real time communication system cannot afford to have network delays on name resolution, because it can compromise the whole communication system operation. In case of network failure or a DNS failure, all MX-ONE names can be resolved by the local DNS.

The MX-ONE local DNS Server shall be a sub-domain (a separate DNS zone) of the company DNS Server (domain.com), for example, mx-one.domain.com.

Alternative 1:

Configure the corporate DNS to delegate the DNS Zone, for example mx-one.domain.com, to MX-ONE Server 1 (DNS master).

The delegation is effected by advertising mx-one.domain.com as a "NS RR" record and an "A RR" record mapping mx-one.domain.com to the IP address of MX-ONE Server 1. The first query to the corporate DNS on some host in mx-one.domain.com will now

be forwarded to DNS master in MX-ONE Server 1. The result will be cached in the corporate DNS in order to lessen the traffic to MX-ONE Server 1.

The master DNS in MX-ONE Server 1 only resolves its own domain hosts, so for external DNS queries, the corporate DNS must be set as the forwarder DNS server.

The figure below explains the interaction when several DNS are involved.

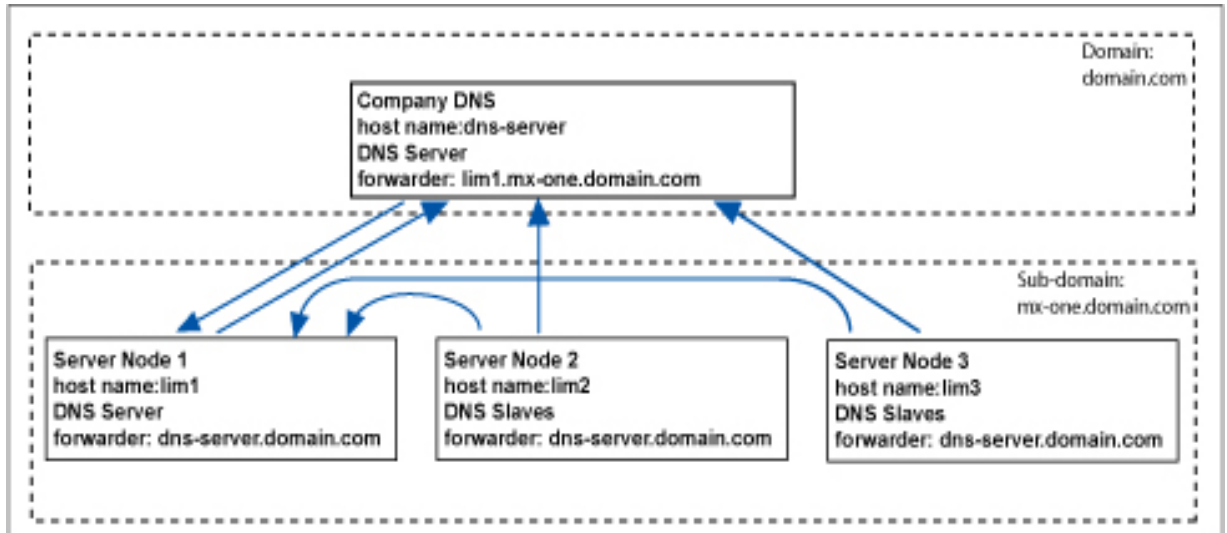


Figure 6: Interaction when several DNS are involved

Alternative 2:

All the hosts resolved by the DNS master in MX-ONE may be defined as a DNS Zone in the corporate DNS instead of defining a delegation to the MX-ONE Server 1. The reason for this could be to ease administration of SRV Records and multiple A records for the mx-one domain. The drawback is that the DNS Server data between DNS master in MX-ONE and the corporate DNS Server may diverge over time.

To be able to register terminals in TLS mode the internal names used by MX-ONE must be added to the DNS master. See operational direction for Certificate Management for description of the internal FQDN name.

3.2.3

NETWORK TIME PROTOCOL (NTP)

Network Time Protocol (NTP) is widely used to synchronize system clocks among a set of distributed time servers and clients.

MX-ONE uses NTP to synchronize system clocks. It uses an external NTP. The NTP server (specified in the configuration file) synchronizes the clocks of all the MX-ONE Service Nodes.

4 TELEPHONY NETWORK PLANNING

4.1 TRUNK PLANNING

When performing trunk planning for PSTN or other public TDM-based network, two types of traffic needs to be considered:

- Traditional links that connect the PBX to the public network, and the capacity needed for this type of traffic.
- The amount of trunks needed for extension services like mobile extension and remote extension.

Note: Some countries or operators might require an inter-working unit, like for example, a Channel Service Unit (CSU) in the USA, to connect the system to the public network.

4.1.1 SIP

Supported SIP trunk scenarios, both as tie-line and as public trunk, are IP communication with SIP-enabled VoIP service providers, least cost routing, single external number and third party registration (for operators that only support SIP extensions).

The SIP solution in MX-ONE uses a Back-to-Back user agent (B2BUA) that acts as a user agent to both ends of a SIP call. The B2BUA is responsible for handling all SIP signaling between both ends of the call, from call establishment to termination.

To SIP clients, the B2BUA acts as a user agent server on one side and as a user agent client on the other (back-to-back) side. The basic implementation of a B2BUA is defined in RFC 3261. For a list of RFCs and ITU-T standards, see the description for *MiVOICE MX-ONE*.

The B2BUA may provide the following functionality:

- Call management (for example, billing, automatic call disconnection, and call transfer)
- Network interworking (perhaps with protocol adaptation)
- Topology hiding (of network internals, for example, private addresses and network topology.)

For this release, no standard exists for a SIP trunk but Mitel is committed to support emerging standards, for example, SIPconnect and IMS, and actively works with major operators.

Note: If UDP packet fragmentation in the network is not allowed, i.e. if the “Don’t fragment” flag is set in the IP headers, or if fragmentation should be avoided, then it is recommended to use TCP as SIP transport. The current version of the SIP stack does not support handling of ICMP error messages and automatic switch-over to TCP.

4.1.2 ISDN DSS1 (PUBLIC TRUNKS)

The trunks available in the MX-ONE are based on ISDN E1 or T1 services. The trunks are mainly intended for connection with PSTN and PLMN networks, but can also be used for private networking (see 4.1.3 ISDN QSIG (private networking) on page 16) as well as for connection to media servers like Conference Units, Speech Recognition servers and similar external systems.

The following is recommended:

- For systems with normal traffic (< 0.15 Erlang), use a maximum of 240 extensions per E1 and 200 extensions per T1.
- For systems with heavy traffic (> 0.15 Erlang), use a maximum of 120 extensions per E1 and 100 extensions per T1.

Note: When using integrated DECT, the accuracy on the incoming synchronization must be better than +/-25 PPM.

4.1.3

ISDN QSIG (PRIVATE NETWORKING)

When the system is connected with other PBXes or when a public operator has the ability of running ISDN signaling over the Q-reference point (QSIG) services, an enhanced set of features is available between the systems.

QSIG is used when the system needs to be interconnected with a foreign PBX, when a solution larger than the current system limit is needed or when a solution benefits of being separated into different administrative domains.

QSIG can be transmitted using either ISDN E1/T1 or using IP over Ethernet as carrier. For more information see the relevant documentation.

By tunneling QSIG over IP, it is possible to provide network services to H.323 trunk lines without having to lease tie-lines.

4.1.4

CAS AND OTHER LEGACY TRUNKS

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

The legacy trunks available in the MX-ONE are compatible with the equivalent Mitel MX-ONE TSW trunks, and must be located in MX-ONE Classic, possibly reusing some TLUs at upgrading, if the trunk function is not supported by the MGU. These trunks are mainly intended for connection with PSTN networks, but some can also be used for private networking (see the Operational Directions for Networking) as well as for connection to certain external systems.

4.1.5

MOBILE EXTENSION TRUNK PLANNING

ME is a service that uses public networks to interconnect the extension and the PBX. For this reason, trunk resources have to be reserved and defined for the mobile extensions in the system. Every external call in to or out from a mobile extension occupies two trunks, one in to the PBX and one out from the PBX. It is recommended that the two trunks are placed in the same media gateway, and that the mobile extension's Home Location Register (HLR) is placed in the same media gateway as the trunks.

Place as many mobile extensions as possible in the same network node, to avoid tromboning and speech path delay.

4.1.5.1

Mobile Operator Services and Agreements

To get better control of costs and better service to the users, it is recommended to establish a Service Level Agreement (SLA) with a mobile operator. Most mobile operators offer services to group the enterprise users together like short number services, closed user groups, common number planning, call routing and similar features.

4.1.5.2 *Mitel Mobile Client (Dual mode)*

Mobile extension solution can also be done via the Mitel Mobile Client Controller using a SIP extension in MX-ONE. In that case the mobile users will be seen as a SIP user to the MX-ONE system. The Mitel Mobile Client Controller are connected to MX-ONE via a SIP trunk and SIP extension. All mobile call media are routed through the Mitel Mobile Client Controller except if the Mitel Mobile Client Controller are configured to use Direct Media for GSM calls (recommended setting in order to reduce end to end voice delay in the Mitel Mobile Client+ solution).

The Mitel Mobile Client Controller requires a number of ports in the external firewall to be opened for communication in order for Mitel Mobile Client mobile to work. If Mitel Mobile Client is used for WiFi calls the enterprise WiFi infrastructure needs to be installed and configured to handle real-time traffic and QoS. A Large enterprise campus with several WiFi hotspots needs a WiFi cell planning for WiFi handover between hotspots. Mitel Mobile Client also has LDAP interface for Directory search. Mitel Mobile Client Controller requires access and configuration to inter-work with the LDAP server. Mitel Mobile Client also supports Chat & Presence via XMPP or SIP SIMPLE. Mitel Mobile Client Controller requires access and configuration to inter-work with the XMPP / SIP SIMPLE server, for example Mitel BluStar server.

4.1.6 MX-ONE INTER-GW MEDIA FOR NON-IP ENDPOINTS

The preferred codec will be negotiated both for IP endpoint connections to non-IP endpoint connections and for non-IP extensions and trunks.

When trunks are involved the preferred codec list, set for Call Admission Control, will be used.

4.2 TRANSMISSION PLANNING

This section describes the basic concepts of transmission planning. The basic concepts are used in the assessment procedure described in the next section, see 4.3 Implementation Assessment on page 33.

4.2.1 GENERAL

IP telephony has become an essential part of media transmission in private networks but it also introduces new challenges. Even familiar impairment such as delay and echo causes more significant impact on the speech quality than in a traditional TDM oriented transmission environment. Thus a successful deployment of MX-ONE, an IP-PBX communication system, requires a careful transmission planning to achieve high level of speech quality.

The E-Model (Ref [1]) is a computational model to use for estimation of the end-to-end voice quality. The transmission planning guideline in this section describes the concept and the use of E-Model and explains important parameters that must be taken into planning consideration. The followings are key quality impairments that have major impact on MX-ONE IP telephony features.

- Speech compression (VoIP vocoder) and transcoding
- Delay, including packet delay variation
- Speech level - sender/receiver/overall loudness ratings
- Level and loss plan (related to loudness rating)

- Packet loss
- Echo impairments deployment

It is important to assess all those transmission parameters listed above studying all connection scenarios and the combined effect is evaluated prior to the system deployment.

The transmission plan must consider the combined effect of all parameters to ensure that MX-ONE achieves the overall level of speech transmission quality as perceived by the user (end-to-end speech quality).

Note that Fax-over-IP may require some specific transmission planning and configuration.

4.2.2

REFERENCES

- ITU-T Rec. G.107: The E-model, a computational model for use in transmission planning.
- ITU-T Rec. G.108: Application of the E-model: A planning guide.
- ETSI ES 201 168 (V1.2.1): Speech processing, Transmission and Quality aspects (STQ); Transmission characteristics of digital Private Branch exchanges (PBXs) for interconnection to private networks, to the public switched network or to IP gateway.
- TIA/EIA TSB116: Voice Quality Recommendations for IP Telephony.
- ETS ES 203038: Speech and multimedia Transmission Quality (STQ); Requirements and tests methods for terminal equipment incorporating a handset when connected to the analogue interface of the PSTN.

Note: Analogue telephones of MX-ONE system comply with TBR 38.

- ETSI TBR 8: Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset terminals.
- ANSI/TIA-810-B: ANSI/TIA-810-B specifies the transmission requirements for narrowband digital telephones.

Note: MX-ONE digital terminals (Digital and IP telephones) comply with TBR 8 (European market) and ANSI/TIA-810-B (North American market).

- ETSI ES 202 020 (V1.4.2): Speech Processing, Transmission and Quality Aspects (STQ); Harmonized Pan-European/North-American approach to loss and level planning for voice gateways to IP based networks.

ETSI ES 202 020 Annex.xls

- ANSI/TIA-912-B: Telecommunications - IP Telephony Equipment - Voice Gateway Transmission Requirements.

TR41.1-04-05-011-TIA464C1-TIA912A-LossPlanDetails.xls

- ANSI/TIA-968-B: Technical Requirements for Connection of Terminal Equipment to the Telephone Network.

4.2.3 SPEECH QUALITY MEASUREMENT

4.2.3.1 MOS - Mean Opinion Score

The mean opinion score (MOS, specified by ITU-T Rec. P.800) measures voice quality used in telephony networks to obtain the human user's view of the quality of the network. MOS is a subjective rating of user satisfaction and based on a large sample of listeners scoring a set of voice samples on a scale of 5, where 1 is "Bad" and 5 is "Excellent".

A MOS value of 4 or higher represents acceptable satisfaction and denotes toll quality, a level typically associated with circuit-switched networks. Table 1 Listening quality scale (MOS) on page 19 shows relation between MOS value (range) and the subjective perception.

Relation between MOS value (subjective perception range) and R-value (objective measurement for quality perception) is described in 4.2.3.3 Relations between R-value and MOS value on page 20.

Table 1 Listening quality scale (MOS)

Quality of the speech	MOS Score
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

4.2.3.2 E-model and R-state

The result of the E-model calculation is a transmission rating factor R-value, which is a measure to express the end-to-end speech transmission quality. For successful speech transmission quality in MX-ONE, any connections with R-values below 70 should be avoided as explained in the following sections.

R-value components

The rating factor R is composed of $R = R_o - I_s - I_d - I_e + A$ where:

- **R_o** represents the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise.
- **I_s** represents a combination of all impairments which occur more or less simultaneously with the voice signal (e.g. non-optimum side tone, quantizing distortion, overall loudness and other impairments).
- **I_d** represents the impairments caused by talker/listener echo and delay (even with perfect echo cancelling),
- **I_e** represents the effective equipment impairment (e.g. low bit rate codecs). It also includes impairment due to packet-losses of random distribution.
- **A** , the advantage factor, allows for compensation of impairment factors when there are other advantages of access to the user. For instance, conventional wired equipment has A-factor value 0 while mobile equipment have A-factor varying from 5 - 20 depending on geographical coverage. G.107 also provides the default values for all input parameters used in the algorithm of the E-model.

4.2.3.3

Relations between R-value and MOS value

The R factor is related to MOS as follows:

Equation for conversion - R-value to MOS value

For $R < 0$: $MOS = 1$

For $0 \leq R \leq 100$: $MOS = 1 + 0.035 \times R + 7 \times 10^{-6} \times R \times (R-60) \times (100-R)$

For $R > 100$: $MOS = 4.5$

Table 2 Relation of User satisfaction rating vs. MOS vs. R-factor on page 20 shows R-value and MOS value range with regard to user satisfaction.

Table 2 Relation of User satisfaction rating vs. MOS vs. R-factor

User Satisfaction	R-factor Value Range	MOS Value
Very satisfied	90 - 100	4.34 - 4.5
Satisfied	80 - 89	4.0 - 4.31
Some user dissatisfied	70 - 79	3.6 - 3.96
Many users dissatisfied	60 - 69	3.1 - 3.55
Nearly all users dissatisfied	50 - 59	2.6 - 3.05

4.2.4

SPEECH COMPRESSION IMPAIRMENT

Choice of vocoder affects l_e value (effective equipment impairment). The E-model measures and quantifies the perceived quality of different codecs. l_e factors of different speech compression algorithms are shown in Table 3 Speech compression impairment factor l_e on page 20. Other impairment values such as packet loss or echo are not included

Table 3 Speech compression impairment factor l_e

Codec	Reference	Bit rate	l_e value
PCM	G.711	64	0
SB-ADPCM	G.722	64	-1
ADPCM	G.726	32	7
CS-ACELP	G.729 A(AB)	8	11
MP-MLQ	G.723.1	6.3	15

Note: G.722 is an ITU-T standard 7 kHz wideband speech codec operating at 48, 56 and 64 kbit/s. and supported by MX-ONE IP phones. MX-ONE IP phones are operating at 64 kb/s rate as it provides the best speech quality.

The l_e value shown for G.722 in Table 3 is retrieved from the equation, $l_e, wb = l_e + 15$ where l_e, wb is effective equipment impairment factor defined for wideband codec. l_e, wb value varies depending on actual bandwidth utilized by G.722 codec as shown below.

G.722 (64)	$l_e, wb = 14$	$l_e = -1$
G.722 (56)	$l_e, wb = 18$	$l_e = 3$
G.722 (48)	$l_e, wb = 24$	$l_e = 9$

Note: G.729A is an extension of G.729 (and compatible), but requires less computational power with marginally reduced speech quality (l_e value 10 for G.729 vs.

11 for G.729A). Another G.729 extension of annex B provides a silence compression method that enables a voice activity detection (VAD) module. Only the term G.729A is used in this planning section but G.729A and G.729AB have the same impairment value ($le = 11$).

Figure 7 on page 21 illustrates impairment characteristics of speech compression with regard to one-way delay but in the absence of any other impairment factors. The comparison shows four popular IP codecs, G.711, G.729A, G.723.1 (MP-MLQ, 6.3 kbit/s) and G.726 (ADPCM).

G.711 codec has $le = 0$ at no delay and set to $R = 94$ as default in G.107. G.722 (64) will be slightly better than G.711 (not shown). G.726 starts at 87, G.729A at 83 ($le = 11$) and G.723.1 at 79 ($le = 15$) with zero one-way delay.

As also shown in table 3, G.723.1 has larger le values (more quality degradation) and, therefore, has smaller tolerance against delay impairment for a given voice quality level (e.g. $R=70$). For this reason, G.723.1 is not recommended and even excluded in capability negotiation in MX-ONE IP-Telephony application.

The shaded region marked "PSTN quality" represents the traditional wireline PSTN and is bounded by the best G.711 performance on the top, $R = 80$ on the bottom and delay between 0 and 100 ms.

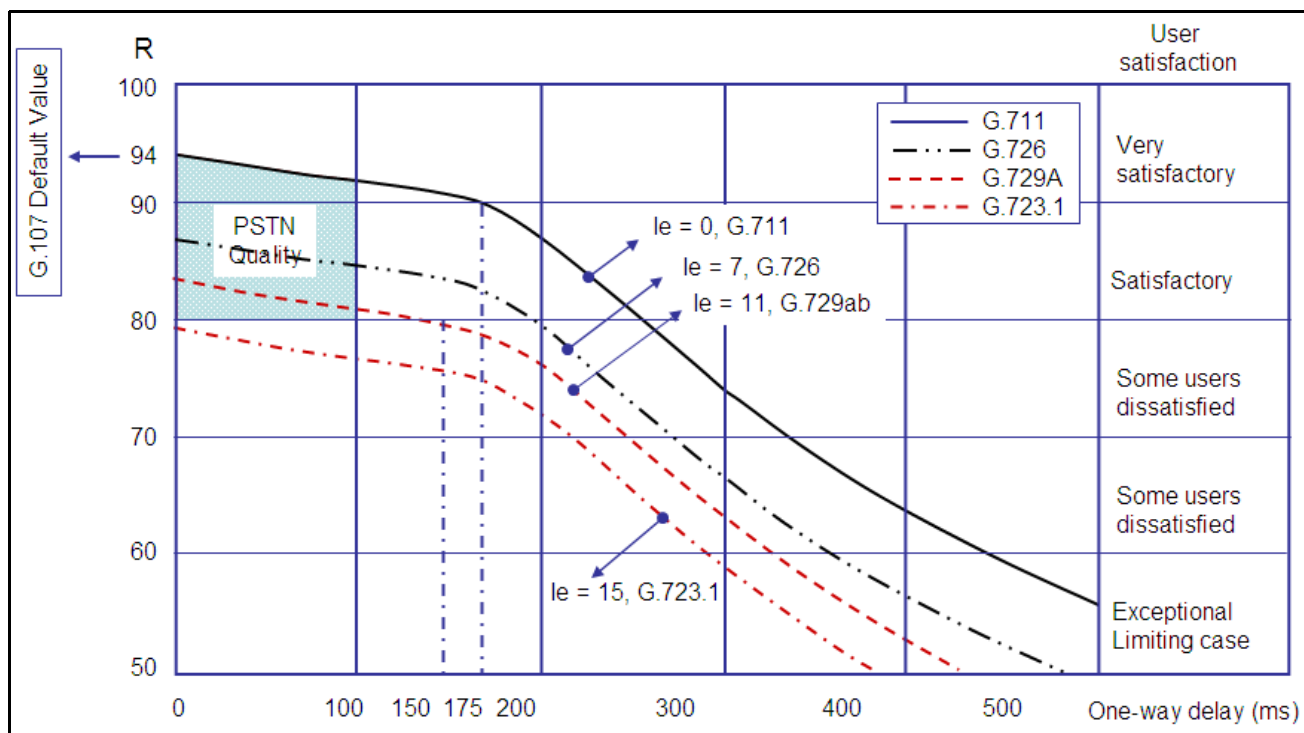


Figure 7: Speech compression impairment with one-way delay

4.2.4.1

Impact of transcoding

Transcoding is defined as two or more encodings of different types of non-G.711 codecs, separated by G.711. For instance, a connection between two DECT portable terminals requires G.726 to G.711 to G.726 transcoding. Even worse, DECT terminal to an IP phone using G.729A codec requires G.726 to G.711 to G.729A transcoding through MX-ONE media gateway.

According to E-Modeling Rule, le -values of speech compression impairment are additive.

For instance, G.726 to G.711 to G.729A results in;

le total = G.726 (le = 7) to G.711 (le = 0) to G.729A (le = 11) = 5 + 11 = 16.

The voice quality degradation due to transcoding is much more complicated issue but this example demonstrates that the i.e factor is severely affected due to transcoding. Generally speaking, only one transcoding can be tolerated before the performance drops below acceptable levels, for most combinations of non-G.711 codecs.

It is important to reduce this additive impairment by considering "Transcoder Free Operation (TFO) or at least keep minimum steps of transcoding by eliminating unnecessary conversion(s).

4.2.5

DELAY IMPAIRMENT

Overall one-way delay, particularly media connected over IP network, consists of:

- Sender packetization delay
- Jitter buffer in the receiver (fixed, absolute or adaptive)
- Transmission delay in the network equipment
- Propagation delay

Although not to be neglected, Transmission/propagation delay normally takes a small portion of delay budget, but should not be neglected when low speed connections are involved.

Sender packetization delay and receiver delay due to jitter buffer are usually main causes of delay impairment for VoIP communication and described in more details below.

4.2.5.1

Sender packetization delay

The following formulas can be used for calculating the minimum and maximum codec-related processing delay:

Minimum packetization delay for high-speed connections = $N \times \text{frame size} + T_{oh}$ (ms)

Maximum packetization delay for low-speed connections = $2N \times \text{frame size} + T_{oh}$ (ms)

Where: N = number of speech frames per packet; speech frame size is in ms;

T_{oh} = look-ahead, PLC, and additional firmware/hardware delay in ms

For instance, when a packet with 3 frames of G.729A vocoder is used in a connection, the minimum sender delay becomes 35 ms ($= 3 \times 10 + 5$) and maximum delay becomes 65 ($= 2 \times 3 \times 10 + 5$ ms) assuming overhead is 5 ms.

4.2.5.2

Receiver's delay

Receiver delay is a sum of decode processing, jitter buffer delay and other overhead.

Normally decoding process and overhead should not exceed 5 ms and, therefore, the receiver delay is largely dependent on jitter buffer size.

Jitter buffer can be sized frame-based or absolute jitter buffer size.

When jitter buffer size is frame-based, the size of the buffer is a multiple of speech-frame size.

It can significantly increase receiver delay if the packet size is large. The reason is that the jitter buffer should normally be two times the packet size for frame-based buffers.

Absolute jitter buffers are sized (fixed) to the maximum delay of the transport caused by PDV - maximum packet delay variation in time. It can reduce receiver delay since the buffer size can be set optimally when the network condition is known, instead of choosing multiple value of frame size.

For instance, 22 ms max PDV can be set to 22 + safety margin instead of 30 ms. Adaptive jitter buffer can be considered as a more enhanced method of absolute jitter buffer.

In summary, the following formulas can be used to estimate receiver delay.

- Frame-based jitter buffer delay = $2N \times \text{frame size (ms)}$, recommended
- Absolute jitter buffer delay = actual end-to-end delay variation + margin (ms)

Where: N = number of speech frames per packet; speech frame size is in ms.

Absolute jitter buffer delay is always smaller than (or equal to) frame-based jitter buffer.

Refer to the "Media Gateway Unit, MGU" description for more technical details.

4.2.5.3

Improvement of delay

From delay impairment point of view, it will result minimum delay to choose minimum packet size containing a single frame and minimum jitter buffer size on receiver. But badly dimensioned buffer size will cause frequent packet loss if PDV, packet delay variation, exceeds buffer size causing that overall speech quality is affected negatively by the combined impairment factor.

It is an issue of balancing minimum delay and the price to pay for other factors working against minimum delay, i.e. more bandwidth requirement and more DSP processing capacity.

4.2.6

LOUDNESS RATINGS

The speech transmission components of terminal elements (analogue, digital, IP or cordless phones) are characterized by Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) which contribute to the Overall Loudness Rating (OLR).

OLR of a connection is expressed as:

$$\text{OLR} = \text{SLR} + \text{CLR} + \text{RLR}$$

CLR, Circuit Loudness Rating, represents the sum of all analogue and digital losses between these telephone sets. An important loss factor of CLR includes level adjustment in a connection such as relative levels (dBr), loss plan and other loss/gain that might be introduced in the connection.

4.2.6.1

MiVoice MX-ONE Terminals

MX-ONE terminal elements comply with following standards.

Analogue phones

ETSI TBR 38 Public Switched Telephone Network (PSTN);

Digital and D4 IP phone

ETSI TBR 8 Integrated Services Digital Network (ISDN) Telephony 3,1 kHz teleservice;

D5 narrow band IP phone

ANSI/TIA-810-B Transmission Requirements for Narrowband Digital Telephones

Table 4 MX-ONE terminals Normal SLR Values

SLR - Send Loudness Rating		
Analogue phone	+3 +/- 4 dB <small>Note 1</small>	(ETSI TBR 38)
Digital and IP phones	+7 +/- 3.5 dB <small>Note 2</small>	(ETSI TBR 8)
	+8 +/- 4 dB <small>Note 2</small>	(TIA 810)

Table 5 MX-ONE terminals Normal RLR Values

RLR - Receive Loudness Rating		
Analogue phone	-8 +/- 4 dB <small>Note 1</small>	(ETSI TBR 38)
Digital and IP phones	+3 +/- 3.5 dB <small>Note 2</small>	(ETSI TBR 8)
	+2 +/- 4 dB <small>Note 2</small>	(TIA 810)

Note 1: At maximum volume setting

Note 2: Normal volume setting

ITU-T Rec. G.107 specifies SLR = 8dB and RLR = 2 dB

4.2.6.2

E-model rating of OLR

The relation between the E-Model Rating R and the OLR of a connection is shown in Figure 8 on page 24. This graph, calculated with the E Model, is obtained when all other input parameters of the E-Model are set to their default values and OLR is the only impairment in the connection considered. The recommended value for the Overall Loudness Rating (OLR) for standard applications of 3.1 kHz handset telephony is 10 dB (ITU-T Recommendation G.111).

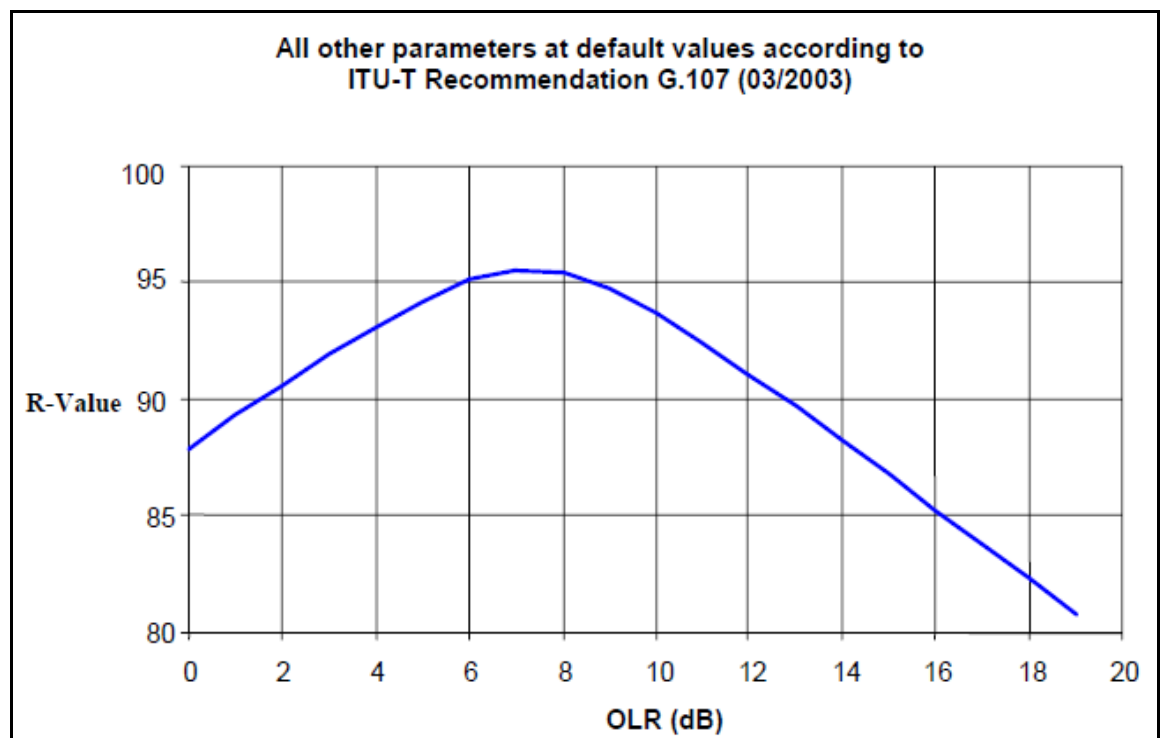
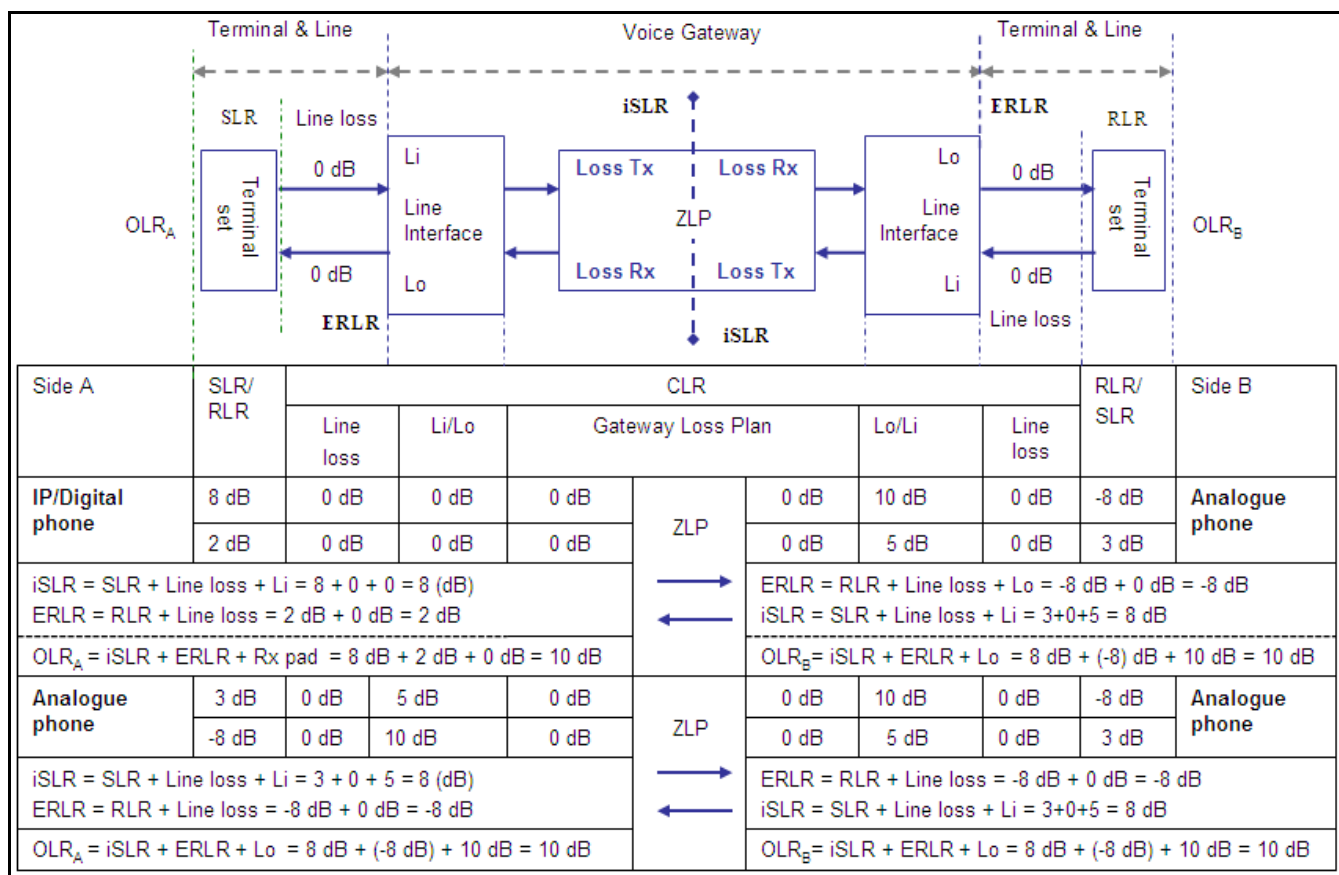


Figure 8: OLR and E-model rating R

4.2.6.3

OLR calculation

As mentioned in 4.2.6.2 E-model rating of OLR on page 24, the internationally recognized and recommended optimum OLR is 10 dB, CLR (circuit loudness rating) is therefore important parameter to be used to achieve a desirable port-to-port OLR value of 10 dB although the optimum value can not always be achieved in all connection types. Figure 10 on page 25 illustrates computation of OLR in several simple connection cases.

**Figure 9: OLR derivation based on ETSI ES 202 020 Annex.xls (Reference 8)**

Note 1: It is assumed SLR = 8 dB, RLR = 2 dB for IP/Digital phone and SLR = 3 dB, RLR = -8 dB for analogue phone.

Note 2: CLR is the sum of all loss values Tx and Rx loss pads in port interfaces and loss values in the gateway. These Tx and Rx losses are 0 dB for digital extension ports. For analogue extension ports (short line), ETSI recommends, 5 dB loss in Tx direction (relative level Li) and 10 dB loss in Rx direction (relative level Lo).

Note 3: iSLR (IP SLR) is an equivalent SLR (ESLR) at the voice gateway-to-IP network connection point (also called ZLP, zero level point) and referred to IP send loudness rating at WAN interface. The iSLR is expected to be 8 dB as it obviously would be the case for IP/digital phone (SLR = 8 dB and all 0 dB loss till ZLP). This assumption, then, applies to analogue phone to determine proper relative level, i.e Li = 5 dB which results in iSLR = 8 dB (= 3 dB + 5 dB).

Note 4: ERLR - Equivalent Receive Loudness Rating, are RLR calculated at certain interface point of voice gateway. Here ERLR is defined at the edge of media gateway (interface ports of digital or analogue extension boards).

4.2.6.4

Loss plan of MiVoice MX-ONE media gateway

The OLR calculation shown in 4.2.6.3 OLR calculation on page 25 assumes that loss values are padded in the port interfaces (i.e. relative level, dBr) to achieve the optimum OLR value. The underlying idea (Reference [8]) is that the necessary loss values are standardized and embedded in overall level plan of interface ports so as to keep voice gateway loss-free.

However, the level plan of MX-ONE analogue interface ports is market dependent, i.e. relative levels are different for different market. For this reason, MX-ONE gateway needs to insert additional loss value unique to each market to obtain the optimum overall loudness rating.

Table 6 MX-ONE loss plan and OLR derivation on page 26 shows an example where Tx loss is 0 dB and Rx loss is 7 dB. It is equivalent to 0 dBr level point in transmit direction (port-to-ZLP) respective -7 dBr level point in receive direction (ZLP-to-port).

Table 6 MX-ONE loss plan and OLR derivation

	SLR	CLR		iSLR	CLR		RLR	OLR
Analog phone		TX pad	Loss plan		Loss plan	RX pad		
	3dB	0dB	5dB <small>(note 1)</small>	8	3dB	7dB	- 8dB	10dB
Note 1: iSLR value is expected to be 8 dB. Hence additional 5 dB loss value is added in loss plan ($3 + 0 + 5 = 8$).								
Note 2: To achieve OLR 10 dB, additional 3 dB loss value is to be added ($-8 + 7 + 3 = 2$ dB) resulting in OLR = 8 dB ($iSLR + 2 \text{ dB} = 10 \text{ dB}$).								

Refer to the document "MARKET CHARACTERISTICS" for more details of market dependent loss and level plan, tone/line characteristics and any other market dependent data.

See also 4.2.9 regarding harmonization of loss and level planning between the Pan-European and the North American regions, ETSI published ES 202 020 (Ref [8]) and an equivalent ANSI/TIA-912-A (Ref [9]) published by the North American Telecommunications Industries Association

4.2.7

PACKET LOSS

Packet Loss is caused by a variety of reasons such as high levels of congestion that lead to buffer overflow in routers (random early detection), link failure, misrouted packet and packet deletion in receiver buffer due to excessive packet delay variation and so on.

It is shown in figure 10 how various packet loss rates affect le (effective equipment impairment) values and the delay margin is significantly reduced. It should be noticed that the figure7 on page 27 in section 4.2.4.1 Impact of transcoding on page 21 is in fact the reference curves for G.711 and G.729Ab with 0% packet loss.

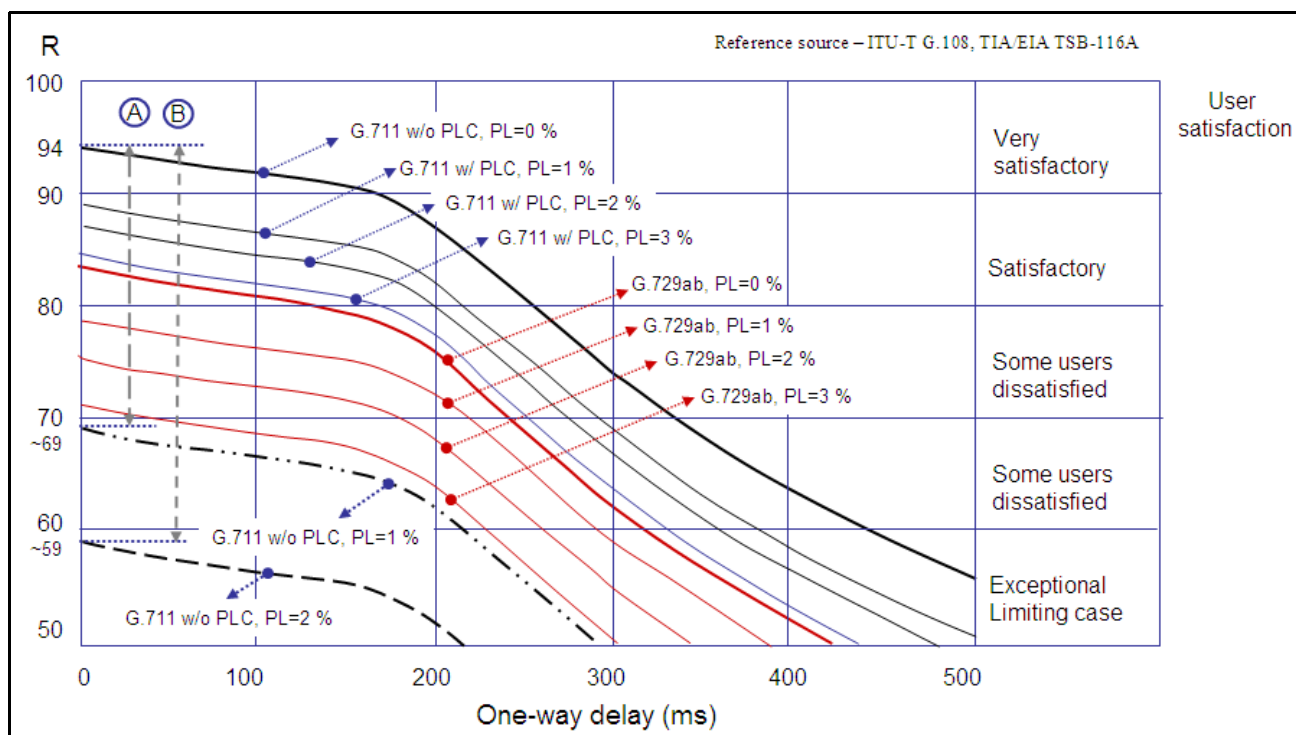


Figure 10: G.711 vs. G.729Ab packet loss performance

G.711 vocoder is much more vulnerable for packet loss. At no delay the *le* value increases about 25 at 1% packet loss rate (marked A) and about 35 at 2% packet loss rate (marked B). In fact, G.711 with 1% packet loss performs worse than G.729Ab with 3% packet loss and is hard to meet the performance level of *R* = 70 which is the ITU-T recommendation of minimum voice quality level for IP telephony.

Vocoders like G.729 and G.723.1 have built-in PLCs and, therefore, less sensitive against packet loss but G.711 (as well as G.726) codec which has been originally intended for switched circuit networks requires PLC add-ons, even at packet loss rate < 1%.

There are two standard methods for G.711 to equip with PLC, G.711 with PLC random packet loss performance and bursty packet loss performance (note).

Note: ANSI T1.521 Annex A PLC algorithm approved by ITU-T as Appendix I of Recommendation G.711

PLC algorithm is estimated to add about 5 ms of processing delay but they are essential in IP telephony as illustrated in figure 10. The additional 5 ms delay should be considered as a very small price to pay.

G.711 with PLC (bursty or random type) should always be prioritized over G.711 without PLC in all VoIP connections.

4.2.8

ECO IMPAIRMENT

4.2.8.1

Terminology

- ECAN** Echo canceller (implemented in DSP of media gateway)
- ERL** Echo Return Loss (Transhybrid loss, echo attenuation at hybrid)

ERLE	Echo Return Loss Enhancement - ability of an echo canceller to suppress echo signals
TELRL	Talker Echo Loudness Rating

4.2.8.2

Cause of echo

The echo is caused from a mixed analog and digital routing within private and public networks, when reflections occur in the presence of 4-wire/ 2-wire conversion. The effects of echo in a conversation can cause impairments to the talker as well as to the listener. These impairments are expressed as Talker Echo and Listener Echo. In general, the listener echo can be neglected (ITU-T Rec. G.131) if there is sufficient control of the talker echo.

4.2.8.3

Estimation of echo

Echo estimation is measured with TELR, talker echo loudness rating, which is perceptual loudness of echo that talker experiences at both end-points. TELR is the sum of the losses around the echo loop, from one terminal element (analogue, digital or IP phone) back to the receiver on the same terminal. Refer to Figure 10 which illustrates different inter-gateway connection types and inclusive parametric values that contribute to TELR.

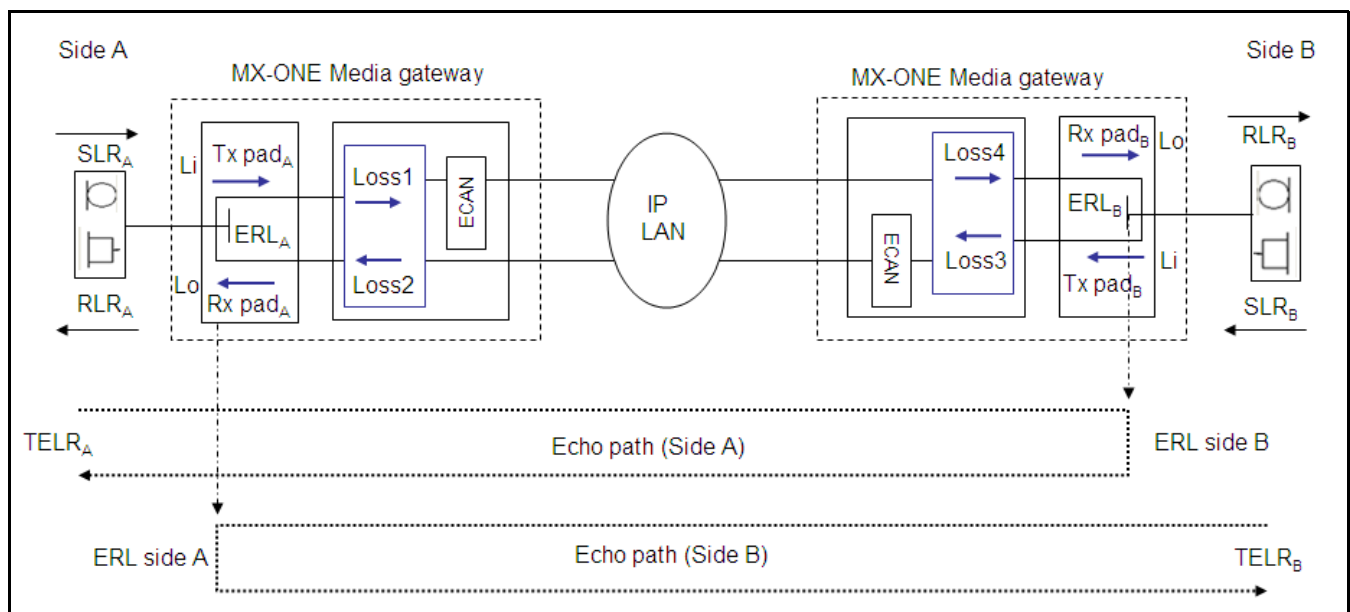


Figure 11: Echo loop and echo estimation

TELR derivation

$$TELRA = SLRA + Tx\ padA + Loss1 + Loss4 + Rx\ padB + ERLB + Tx\ padB + Loss3 + ECAN + Loss2 + Rx\ padA + RLRA$$

$$TELRB = SLRB + Tx\ padB + Loss3 + Loss2 + Rx\ padA + ERLA + Tx\ padA + Loss1 + ECAN + Loss4 + Rx\ padB + RLRB$$
TELRA estimation

Note: TELRB is estimated in a similar way.

Case 1 - IP/digital phone on side A and analogue telephone on side B:

Table 7 Case 1 parametric values for echo estimation

Side A	
$SLR_A = 8 \text{ dB}$, $RLR_A = 2 \text{ dB}$	Default SLR and RLR of IP/Digital phone
Loss 1 = Loss 2 = 0 dB	Digital connection
$Tx \text{ pad}_A$, $Rx \text{ pad}_A = 0 \text{ dB}$	Digital ports
Side B	
$SLR_B = 3 \text{ dB}$, $RLR_B = -8 \text{ dB}$	Default SLR and RLR of analogue phone
Loss 3 + $Tx \text{ pad}_B = 5 \text{ dB}$	Default loss pad for analogue Tx port = 5 dB
Loss 4 + $Rx \text{ pad}_B = 10 \text{ dB}$	Default loss pad for analogue Tx port = 10 dB

Case 2 - Analogue telephone on both side A and side B.

Table 8 Case 2 parametric values for echo estimation

Side A	
$SLR_A = 3 \text{ dB}$, $RLR_A = -8 \text{ dB}$	Default SLR and RLR of analogue phone
Loss 1+ $Tx \text{ pad}_A = 5 \text{ dB}$	Default loss pad for analogue Tx port = 5 dB
Loss 2+ $Rx \text{ pad}_A = 10 \text{ dB}$	Default loss pad for analogue Tx port = 10 dB
Side B	
$SLR_B = 3 \text{ dB}$, $RLR_B = -8 \text{ dB}$	Default SLR and RLR of analogue phone
Loss 3 + $Tx \text{ pad}_B = 5 \text{ dB}$	Default loss pad for analogue Tx port = 5 dB
Loss 4 + $Rx \text{ pad}_B = 10 \text{ dB}$	Default loss pad for analogue Tx port = 10 dB

Replacing those values in $TELRA_A$ equation;

$$TELRA = 3 + 5 + 10 + ERLB + 5 + ECAN + 10 + (-8) = 25 + ERLB + ECAN$$

The result is the same as case 1 (which is supposed to be when OLR value is set to 10 dB for both connection cases).

Echo suppression assessment

Assuming the convolution processor part of the echo canceler (in DSP) provides about 18 dB of echo return loss enhancement (ERLE);

- $TELRA > 65 \text{ dB}$ requires ERL better than 22 dB
- $TELRA > 60 \text{ dB}$ requires ERL better than 17 dB
- $TELRA > 55 \text{ dB}$ requires ERL better than 12 dB

The nonlinear processor (NLP) provides an additional loss of at least 25 dB, and assuming typical ERL of 12 dB and ERLE of 18 dB, a total of 55 dB can be obtained as recommended by ITU-T requirement.

More details about the handling and operation of DSP echo canceler (CP and NLP) can be found in the "Media Gateway Unit, MGU" description.

4.2.8.4

E-model prediction of echo impairment

Table 5 and figure 11 show E-Model prediction of echo effect (in reference G.711 codec).

R-value is affected differently by delay value for a given TELR curve and a connection.

Table 9 TELR and delay impairment and resulting R-value

TELR (dB)	65/55/45	65			55			45		
Delay (ms)	0	50	100	150	50	100	150	50	100	150
R-value	94	93	92	91,5	91	86	83	83	71	63

Figure 12 on page 30 illustrates impacts of delay on various total echo suppression measured in TELR. The delay is a crucial factor when a connection is made over IP network since the VoIP connection always operates in long delay environment.

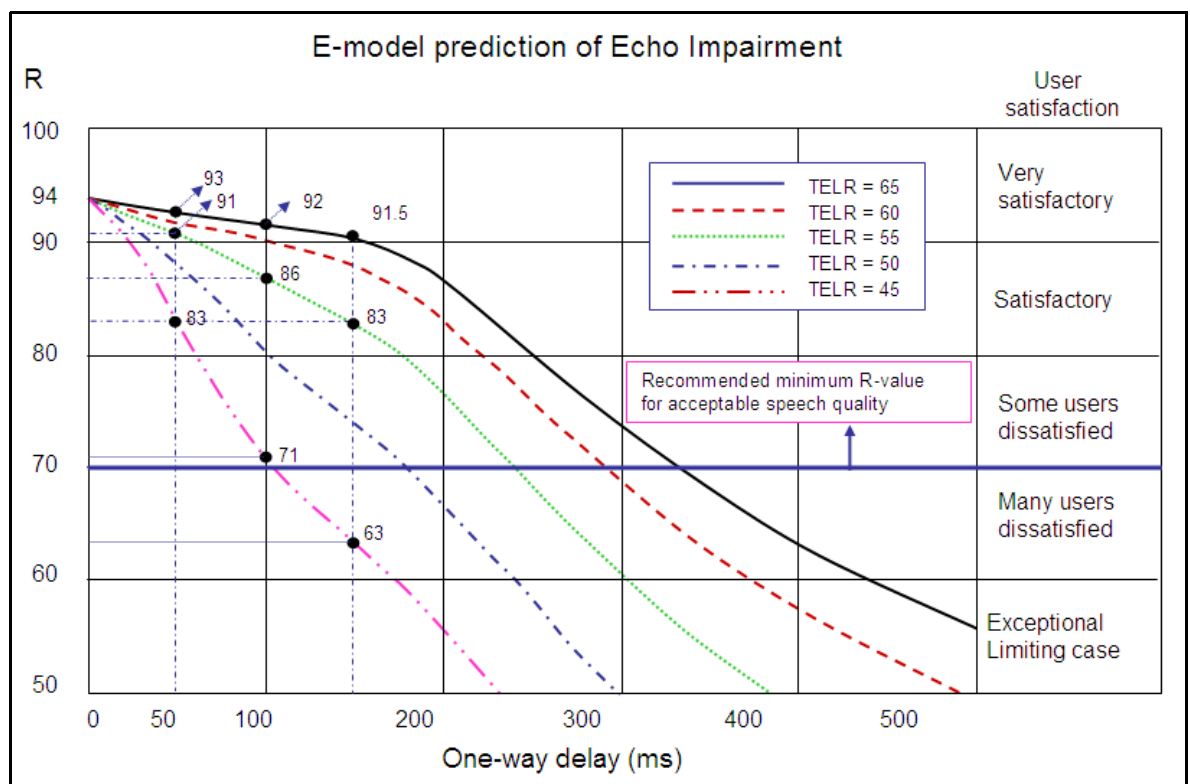


Figure 12: E-model prediction of echo impairment

Improvements in perceived quality with respect to talker echo can either be achieved by higher values of TELR or by lower values of mean one-way delay (T), or by a combination of both. The value of TELR can be increased by improvements in the 4- to 2-wire conversion or by increasing the echo loss in the connection.

It must be mentioned the loss plan must take all the factors into the consideration for all connection types, analogue as well as mixed analogue/digital. It is crucial to provide enough loss to attenuate the echo and maintain circuit stability still being audible over a range of analog loops. However, it is also important to avoid excessive OLR value under a certain given conditions otherwise the excessive OLR value also contributes loudness impairment (section 4.2.6.2 E-model rating of OLR on page 24), also resulting in poor speech quality.

TELR estimation for connection examples shown in 4.2.8.4 E-model prediction of echo impairment on page 30 is based on G.711. Normally, TELR value is expected to be

better than 55 ~ 60 dB with G.711 and with one-way delay less than 100 ms, the worst case echo impairment, I_d will be < 8 (R -value > 86).

The default R -value of G.729Ab is ~ 83 ($I_e = 11$) and the diagram shows that I_d factor becomes 8 ($= 94 - 86$) for TELR value 55 dB and 14 ($= 94 - 80$) for TELR value 50 at 100 ms one-way delay. It implies that the combined R -rating drops to 75 ($= 83 - 8$) respective to 69 ($= 83 - 14$).

It demonstrates how the impact of echo impairment is increased by degraded echo performance (lower TELR) and increasing delay time. Transmission planning must consider this issue to achieve high TELR value (good overall echo suppression) and at the same time reduce one-way delay time as much as possible. For instance, selection of short packet length to reduce sender/receiver delay must be weighed in echo and delay critical connections.

4.2.9

FAX AND MODEM TRAFFIC CONSIDERATIONS

MX-ONE supports Fax over IP (FoIP) based on either T.38 fax relay or "clear channel"/G.711 fax pass-through. There are however a number of limitations to consider when fax shall be transferred over an IP network. The overall performance is highly dependent on the quality of the IP network i.e. the network impairments; packet loss, latency and jitter.

The different fax transport methods put different requirements on the network for reliable operation and performance.

With the T.38 fax relay method, defined by ITU-T, fax data are transmitted directly over an IP network without first being converted to an audio stream, which reduces significantly the required bandwidth. In a mixed packet based and circuit switch network transcoding has to be done between the different carrier methods causing extra latency which will have a negative impact on the overall operation.

With G.711 fax pass-through, the fax data is converted into a Pulse Code Modulation (PCM) audio stream which is sent as G.711 RTP packets.

G.711, defined by ITU-T, is optimized for transmission of voice in an IP network and any packet loss will not have a major impact on the voice quality but when transmitting fax data any loss of packets will have a significant impact on the overall operation.

4.2.9.1

Packet loss in IP network

T.38 has good protection against single packet loss and burst packet loss, especially with redundancy enabled i.e. FoIP operation in an IP network with several percent of packet loss will still give good fax operation.

G.711 has poor protection against both single packet loss and burst packet loss. Already at packet loss above 0,25% the fax operation is reduced noticeable and it can take very long time to transfer a document between two fax machines over the IP network.

4.2.9.2

Latency in IP network

Latency can vary a lot depending on network configuration, number of routers and transcodings etc.

Both T.38 and G.711 give similar performance and generally FoIP operates well with round trip delays less than 2 sec.

4.2.9.3

Jitter in IP network

Jitter is the variation of latency the network adds to transmitted packets. If a packet is coming too late to receiving side, due to network latency, it will be discarded i.e. there is a packet loss.

To reduce the sensitivity to jitter latencies, jitter buffers have been introduced in the network elements and as well in the IP PBX'es. Multiple jitter buffers will though introduce extra latency to the network.

4.2.9.4

Conclusions

While faxes and modems operate in real time they require that data are transmitted and received in a continuous stream. The overall fax performance is heavily depending on the network topology it will be transmitted over i.e. IP, circuit switched or a mixture of both. In a poorly maintained IP network or in network with many transcodings FoIP may not work at all.

MX-ONE is an IP based system where GW nodes are interconnected over an IP network. In order to get maximum fax performance with e.g. an analog fax it might be required to have the fax machine connected to the same GW node where all incoming/outgoing ISDN lines are connected. The analog interface used for fax could also be configured to use bearer capability 3.1 kHz audio (which gives "clear channel", i.e. bit transparency), and/or to use bearer capability conversion for fax traffic on incoming trunks, for better fax performance.

When migrating a circuit switched system, like MD110/Mitel TSW, to an IP switched MX-ONE it is especially important to replan the fax and modem set up in the new system with respect to the impairments coming with an IP based system.

How to configure the MGU board to interwork with fax or modem traffic please consult the descriptions for *MGU Media Gateway Unit* and *MGU2 Media Gateway Unit*.

4.2.10

HARMONIZED PAN-EUROPEAN/NORTH-AMERICAN LOSS AND LEVEL PLAN

There are two standards addressing the loss plans specified for voice gateways in pan-European respective North-American region (listed below). These standards define performance and technical criteria that interface and connection elements of private and public telecommunication networks should comply to assure overall speech quality.

Each standard provides Annex Spreadsheets which contain many practical connection scenarios and calculations based on optimum OLR criteria to reach respective loss plan applicable to each region.

- ETSI ES 202 020 V1.3.1: Speech Processing, Transmission and Quality Aspects (STQ); Harmonized Pan-European/North-American approach to loss and level planning for voice gateways to IP based networks
- ES 202 020 V1-3-1 Annex.xls
- TIA-912: Telecommunications IP Telephony Equipment Voice Gateway Transmission Requirements.
- TR41.1-04-05-011-TIA464C1-TIA9122A-LossPlanDetails.xls

4.3

IMPLEMENTATION ASSESSMENT

The implementation procedure are divided in the following steps:

1. Planning
2. Calculation
3. Deployment

4.3.1

PLANNING

Investigate the following four main areas:

1) **Telephony usage**

- Number of users
- Number of directory numbers
- Extension type distribution (IP, mobile, analog, digital)
- Number of calls
- Duration of calls
- Number of concurrent calls
- Call volume statistics - Peak and average
- Call flow and location - internal versus external calls

This information is usually recorded in the current PBX.

2) **Reliability**

Users expect a high level of reliability for the telephony system. PSTN and PBX is usually calculated to have an availability of 99.999%, that is, system is down during a maximum of 5 minutes per year.

In general, data networks have not reached that level yet. Computer systems on average have been calculated to have a reliability of 98.5%, that is, system is down 5.5 days per year. Most network outages stem from performance issues, like insufficient bandwidth or high peak load. Some security problems like Denial of Service (DoS) attacks add to the problem.

Good network design can eliminate problems with single point of failure. Although a single point design may not fail, it can be the bottleneck in the system.

3) **Call Quality**

Even though they are carried on the same network, the network cannot handle different media types in the same way.

Different media types have different packet sizes, send rates, and demands on reliable delivery. If it takes 5 - 10 seconds to access a Web page or an e-mail it is accepted by users, but if an interactive voice call is delayed more than 150 - 200 ms it is perceived as disturbing by users.

On the other hand, voice calls only use a small amount of bandwidth (usually 20 to 200 kbit/s), while file downloading attempts to occupy the entire available bandwidth.

4) **Network Configuration Assessment**

This assessment investigates the current state of the network equipment to determine if the network is able to carry VoIP traffic.

Before deploying a VoIP solution, do the necessary measurements and simulation of expected traffic in the production network to confirm that no problems or unexpected events occur. There are a number of available tools for VoIP network assessments and this service can be offered by Mitel partners.

The collected data should contain the following information:

QoS – Do the network devices support QoS mechanisms? Is QoS already enabled in the routers and switches? How will VoIP traffic be prioritized?

VLAN – A virtual LAN allows to group or segregate LAN traffic. It also allows prioritizing different data classes by the switches. Do the switches support VLANs and IEEE 802.1p/q?

Shared Hubs – are not supported.

Interface speeds – Which speeds are the different routers running on? Do they support full duplex? Does the interface support the estimated added amount of VoIP traffic?

Operating systems – What version of the operating system is running on the routers, switches, firewalls and other devices? Does it support VoIP or does it need to be upgraded?

Memory – How much Random Access Memory (RAM) is installed in the network device (for example, routers)? Is it enough to support the voice call traffic added to the network?

4.3.2

CALCULATION

The goal of the calculations is to verify if the current infrastructure can support the combined load from existing data and telephony usage.

To make such calculations, it is necessary to identify where the main flows are and where it is most likely that congestion will occur. The following sections identifies a number of issues.

4.3.2.1

Server - Server signalling

When two or more Servers are used, the Servers communicate using the proprietary inter-Server signaling. For information on how to calculate for inter-Server traffic, see 3.1.3.2 Inter-Server Modeling on page 12.

When the solution needs an even larger system, each system node must be interconnected with the SIP networking feature, or use ISDN QSIG (or H.323) networking.

4.3.2.2

Trunk Lines

To calculate the necessary trunk line needs, base the calculations on the assumption of 0.8 Erlang load per trunk.

4.3.2.3

Extensions

It is assumed that a typical user is loading the system with 0.2 Erlang of traffic, during Busy Hour, that is, the user is making 8 calls/hour and each call is in average 90 seconds long. If the business is very call intensive or each call have higher average holding times, extra consideration needs to be taken.

In business situations which have operations going on for 24 hours and 7 days a week, many models assume that a working day is 8 hours and that a week consists of 5 working days. In worst case this could give an error by the factor 4 for average calls/seconds.

4.3.2.4

Bandwidth Calculation

The simplest type of modeling uses the projected call volume where selected codec and bandwidth are required as input parameters. Modeling a whole network will require much computation with different input values and variables. However bandwidth modeling may be the most valuable way for calculation of critical links.

The bit rate stated for a codec is a net value, that is, this value does not include the overhead caused by packet headers. So the real bit rate of a voice stream encoded by these codecs is higher than the bit rate, see Table 10 Supported Audio Codecs on page 35.

Table 10 Supported Audio Codecs

CODEC	Bandwidth (kbit/s)
G.711 A-law, 64 kbit/s	80
G.711 μ -law, 64 kbit/s	80
Clear channel	80
Unknown codec	80 (default)
G.729 annex A and annex B	24
G.729 annex A	24
G.729 annex A	24
G.723.1	17
G.723.1 Annex C	17

To estimate the real bit rate, this value has to be combined with the header bandwidth, see Table 11 IP Layer Bandwidth on page 35 which states the bandwidth needed for different transport media.

Table 11 IP Layer Bandwidth

IP Layer Bandwidth		Inter-Packet Delay			
		10 ms	20 ms	30 ms	
Codec	See 10 Supported Audio Codecs on page 35	-	-	-	kbit/s
RTP	(RFC 3550)	9.6	4.8	3.2	kbit/s
UDP		6.4	3.2	2.1	kbit/s
IP		16	8	5.4	kbit/s
---Total---	RTP+UDP+IP	32	16	10.7	kbit/s

IP Layer Bandwidth		Inter-Packet Delay			
		10 ms	20 ms	30 ms	
- or -	cRTP (RFC 2058)	1.6	0.8	0.5	kbit/s
- or -	cRTP + UDP checksum	3.2	1.6	1.1	kbit/s

To calculate the total bandwidth consumption, it may be necessary to also include link layer bandwidth in the calculation.

In most cases the link layer can be omitted, but with low bandwidth codecs running on narrow band links it is necessary to include additional bandwidth to get a proper result.

Trunk bandwidth = Codec BW * A / A_n, where A = n*α* R_h and A_n is the allowed load on a given trunk. (For values on A_n see 17 Default and Guiding Values on page 72.)

Example: 2 Geographical locations, 160 VoIP clients are needed. G.711 is used (80kbit/s). Current Mean Traffic is 0.15 Erlang per user. Current Mean Hold Time = 67 seconds 2 locations with 80 terminals at each location=> each user needs 80 kbit/s to make a call. The IP trunk (network) needs (80x0.15x1.2x0.5x0.5) = 3.6 Erlang 3.6 Erlang requires 9 channel, thus 9*80 kbit/s = 720 kbit/s is needed. The PSTN trunk carries 7.2 Erlang, and needs 14 channels with GoS=1% (The Hold Time only gives an indication that the call attempts λ are 0.43 calls/s in average.)

It should be noted that the bandwidth calculation assumes a full duplex network end-to-end. On half duplex links the required bandwidth will be twice the value calculated here. Examples for half duplex networks are WLAN and Ethernet hubs.

The next calculation example describes a detailed calculation of the overhead caused by an Ethernet link.

Example: 20 users are using their phones 20 times per workday (8 hours) and each call has an average length of 1.5 minutes. This gives 20 x 20 x 1.5 / (8 x 60) = 1.25 Erlang. 1.25 Erlang of traffic is carried by 6 channels with a probability of 0.2% for lost or congested calls, according to the Erlang Loss Formula table.

The codec bandwidth is calculated from the sum of (MAC layer + IP header + UDP + RTP + Payload) / packets per second. A voice channel that uses G.711 as codec and is sent over an IP network in packets of 30 ms samples, requires 240 bytes payload, 40 bytes IP/UDP/RTP header and 18 bytes of Layer 2 header. This gives (240 + 40 + 18) x 8 bits/byte / 0.03 seconds = 79.47 kbps. If G.729 is used with the same packet, it requires 23.47 kbps.

6 channels would require 6 x 79.47 = 476.8 kbps if G.711 with a packet size of 30 ms is used as codec and 6 x 23.47 = 140.8 kbps if G.729 with the same packet size is used.

476.8 kbps gives that 20 users will add approximately 5% traffic load to a 10 Mbps connection. (Or 10% if the link is not full duplex)

The modeling gets very quickly complicated as parameters like Silence Suppression, Header Compression and Quality of Service are added. The recommendation is to start the deployment calculations from a simple view.

4.3.3

DEPLOYMENT

From the planning phase, the following data and information should have been collected:

- Number of additional IP addresses needed
- Link bandwidth on MAN/WAN connections
- Number of TDM trunk lines needed

QoS - It is assumed that all network devices support QoS mechanisms and that QoS is possible to enable in the routers and switches. It is of great importance that VoIP traffic is given priority to other traffic, and that voice packets are discarded in a controlled fashion.

VLAN - VLAN allows LAN traffic to group or segregate. It also allows prioritizing different data classes by switches. Do the switches support VLANs and IEEE 802.1p/q?

Shared Hubs / Repeater - Shared Hubs do not offer QoS. Verify that device attached to a repeater/hub not is intended to use for VoIP services.

4.3.4

SECURITY CONSIDERATIONS

Most of the concepts that have been illustrated in the previous sections have a significant impact on the security of the whole Voice over IP (VoIP) network.

The use of VLAN, for example, makes the telephony communication occurring over the LAN more resilient to DoS attacks. Most of these attacks originate from PCs in the LAN. Therefore PCs should be located in a different VLAN than the VoIP terminals. Then only the VLAN where the DoS attack originated will be affected. The separation of different types of traffic improves the security of the network as it is able to limit the widespread of an attack and facilitates diagnostic actions.

The use of firewalls and Intrusion Detection Systems (IDS) provides an additional degree of security to the network. However, it is necessary to use devices that are aware of the application layer protocol exchanges, as H.323 and RTP for example, use dynamically-allocated ports that are negotiated during the set-up of the VoIP connections.

The proprietary inter-Server signaling protocol is used for management messages, control signals, and call control signals.

For more information, see installation instructions for *CERTIFICATE MANAGEMENT*.

It is highly recommended to use Virtual Private Network (VPN) when having a VoIP network composed of physically separated subnets and public networks are being traversed.

For more information about Security, see the description for *SECURITY* and the user guide *SECURITY GUIDELINES*.

4.3.5

GUIDELINES FOR SPEECH QUALITY

To achieve good speech quality on an IP network certain rules have to be followed. The main rules are that:

- The total end-to-end one-way delay should be kept below 150 ms. One-way delay = Packetization delay + Propagation delay + Transport delay + De-Jitter Buffer delay.
 - **Packetization delay** is the fixed time needed by the transmitter to gather audio samples, put them through a codec and placing the right amount of codec frames into an IP packet.
 - **Propagation delay** is the time it takes to travel the physical distance from one point to another. When traffic has to cover long distances, be sure that the network path is optimized and as short as possible.
 - **Transport delay** is the total time the packet has to spend in the different devices making up the network, like switches, routers, traffic shapers, gateways, bridges, and so on. Some devices add more latency than

others. Look for the number of hops needed for a packet to go from point A to point B. Try to focus on the worst offenders and try to avoid or replace them.

- **De-jitter Buffer delay** is used to take care of the variations of packet arrival caused by the network. The delay in the buffer can be long, if jitter is high and network delay is low, but this is the point where **trade-off between delay and packet loss** has to be considered.
- The transmitters typically have a delay of 20 - 30 ms, jitter buffers typically delays by 40 - 60 ms and PSTN has a typical propagation delay ranging from 10 - 30 (maximum 50 - 70 ms). This gives some **30 - 80 ms delay left to the IP network**, when the total delay budget should be lower than 150 ms. **The recommendation is to design the IP network with less than 50 ms of delay.**
- Packet loss should be kept below 2% as this has been shown to be a level after which users start to complain of poor voice quality. For Fax-over-IP 2% is not acceptable. Also try to avoid bursts of consecutive packet loss. Increasing bandwidth and good tuning usually solves the problem. It is recommended to keep the **target to less than 1% loss** when designing the network.
- The bandwidth required for the designed number of voice calls should be allocated so that it is available at least 90% of the time. (This means that all other traffic should be limited by traffic policies.)
 - Avoid slow speed links, extensively. Plan to upgrade bandwidth on paths where different media types should co-exist.
 - Use data packet fragmentation for low speed links.
 - Use RTP header compression for low speed links.
- When Traffic Shaping is applied, apply at least 10% over-provisioning to voice calculation. Some situations though might require higher over-provisioning ratio to work properly. It is recommended that Low Latency Queuing (LLQ) or Weighted Fair Queuing (WFQ) is the preferred choice of queuing used for the voice channels. Other methods like Random Early Discarding (RED) will further help up to avoid continuous packet loss. Be aware of that the typical behavior of shapers is that they usually throw away UDP packets first, as they are not resent. Will the traffic shaper affect RTP packets?

For more information see the ITU-T Standards G.107, G.113 and H.460.9, the description for QUALITY OF SERVICE and (if H.323 end-points are used) also the description for CALL INFORMATION LOGGING, QUALITY OF SERVICE LOGGING.

4.3.5.1

VPN Links

Speech quality on a VPN link depends on the operators network and how the company is interconnected with that network. Consider how the current SLA covers putting voice data there.

4.3.5.2

Network Address Translation

MX-ONE Service Node has no mechanism for resolving VoIP signaling (SIP/H.323) via Network Address Translation (NAT). NAT typically occurs when signaling passes firewalls.

If a corporate firewall is needed it is recommended to use Application Layer Gateways (ALG) as firewall (also known as Session Border Gateway(SBC)). The ALG's main

function is to represent the external party or show a path to the external party using ports and addresses that MX-ONE Service Node can access, i.e. the inside of the ALG. For SIP, MX-ONE supports the headers Path, Route and Record-Route according to RFC5626 (Managing Client-Initiated Connections in SIP) which is the path to the external party

The ALG should support:

- dynamically open and close media ports negotiated in the signaling.
- remote NAT traversal, which is to keep up a signaling tcp session to devices (IP phones) residing behind NAT firewalls like for example a home network in a tele-worker solution.

4.4 DECT SYSTEM PLANNING

There are three DECT solutions available for the system; SIP-DECT, IP-DECT and integrated DECT.

4.4.1 SIP-DECT SYSTEM PLANNING

This section describes the planning required before implementing a SIP-DECT solution in an MX-ONE system.

The following needs to be considered when designing an SIP-DECT network:

- To achieve optimal performance the infrastructure should be connected to a switched network (no hubs or repeaters).
- When setting up a network supporting different media types, it is recommended that voice and data are separated on different VLANs.
- Maximum load of the VoIP traffic must not exceed 25% of the capacity of the network.
- Maximum load of the network must not exceed 75% of the total capacity of the network, including the VoIP traffic.
- It is not recommended to use firewalls in the network. But if firewalls needs to be used, tunneling or application aware firewalls should be used.
- Depending on network size, a backbone of at least 100 Mbps should be used.
- The Master OMM registers to one MX-ONE Service Node and all users will be connected to that Server as SIP extensions.

For more information, see the *SIP-DECT - Installation, Administration and Maintenance*.

4.4.1.1 Synchronization Configuration

Each SIP-DECT Base Station (RFPs) must be placed within the air synchronization coverage of at least one other base station. Since the air synchronization cover radius is somewhat larger than the speech radius it is often enough to consider the speech coverage when placing the base stations.

Always ensure that all RFPs are able to synchronize with, preferably two, but at least one other base station.

4.4.2

IP-DECT SYSTEM PLANNING

This section describes the planning required before implementing an IP-DECT solution in an MX-ONE system. The IP-DECT solution can use both H.323 and SIP signalling.

The following needs to be considered when designing an IP-DECT network:

- To achieve optimal performance the infrastructure should be connected to a switched network (no hubs or repeaters).
- When setting up a network supporting different media types, it is recommended that voice and data are separated on different VLANs.
- Maximum load of the VoIP traffic must not exceed 25% of the capacity of the network.
- Maximum load of the network must not exceed 75% of the total capacity of the network, including the VoIP traffic.
- It is not recommended to use firewalls in the network. But if firewalls needs to be used, tunneling or application aware firewalls should be used.
- Depending on network size, a backbone of at least 100 Mbps should be used.
- The Master Server registers to one MX-ONE Service Node and all users will be connected to that Server as SIP extensions.

For more information, see the document *IP-DECT System Planning*.

4.4.2.1

Synchronization Configuration

Each IP-DECT Base Station (called IPBS in this solution, but same as RFP) must be placed within the air synchronization coverage of at least one other base station. Since the air synchronization cover radius is somewhat larger than the speech radius it is often enough to consider the speech coverage when placing the base stations.

Always ensure that all IPBS are able to synchronize with, preferably two, but at least one other base station.

4.4.3

INTEGRATED DECT SYSTEM PLANNING

This section describes the planning required before implementing an integrated DECT solution in an MX-ONE system.

4.4.3.1

Radio Planning

The main factors that determine the final configuration of a system are:

- Number of base stations
- Number of cordless phones
- Traffic requirements for the system
- Physical nature of the site

There are also a number of restrictions that can limit the DECT system configuration, such as, the hardware limitations of boards, base stations, cordless phones and cabling. To create a DECT system configuration, all limitations of the system must first be known.

The traffic requirements for the system are determined by the number of cordless phones needed, the estimated traffic generated per portable and the Grade Of Service (GOS) accepted by the customer.

A major task when planning a cordless phone system is to define the number of base stations required to cover an area to a satisfactory level, that is to gain full area coverage.

The location of the base stations are determined by the following factors:

- Coverage
- Traffic
- Capacity

Another aspect of the base station configuration is the power supply, which is explained in detail in the installation instructions for *INTEGRATED DECT*.

For detailed information on DECT system planning, see installation planning for *INTEGRATED DECT*.

4.4.3.2

Synchronization Configuration for DECT

In MX-ONE each media gateway is a synchronization node. This means that all media gateways that are equipped with ELU31 boards must receive synchronization from a common external (or internal) synchronization source. This synchronization source could be an E1/T1/BRI or, if not available, an internal clock source on MGU board can be used as synchronization source. The GW node connected to the common synchronization source will be called the Master sync GW. Master node is then distributing a system synchronization with a stability of better than +/-5ppm to all other GW nodes, containing DECT boards, via the DECT synchronization ring.

Disturbances in the external synchronization or in the synchronization ring can cause major problems in a DECT system. Therefore it is very important that the synchronization functions is carefully planned and according to the documentation.

For more information, see the installation instruction for *INTEGRATED DECT*.

4.4.3.3

SMS Configuration for DECT

In MX-ONE it is only possible to initiate one SMS server per MX-ONE Service Node. The server should always be initiated with the MX-ONE Service Node IP address. To decrease inter-Server signaling it is recommended to always initiate the SMS server in a MX-ONE Service Node to an MX-ONE Classic, where DECT boards are also located. The (generic extension) directory number of the SMS server should be initiated with the 'SMS permitted' parameter.

It is recommended to assign the IP addresses to the same subnet as the IMS servers.

Another aspect of SMS configuration is the Integrated Message Server, IMS, server configuration, which is described in the IMS manual provided with the IMS hardware.

4.5

WLAN PLANNING

It is recommended to use the WLAN manufacturer's installation guide for system planning, logical connection, and configuration of the WLAN system and APs.

For more information on single VoWiFi system, see the description for *WI-FI SYSTEM PLANNING*.

4.6 NUMBERING

4.6.1 NUMBERING PLAN

When planning the numbering plan for the site to be installed, it is important to examine the existing telecom environment and what the customer requires.

This section only mentions the most important items to think about.

For detailed information about numbering see operational directions for *NUMBERING*.

4.6.1.1 *Public Numbering plan*

Direct in-dialing requires that a number range in public network have to be subscribed from the Telecom operator if not already available. Make sure that the subscribed public number range is sufficient for the present installation and planned expansion.

4.6.1.2 *Internal Numbering plan*

If direct in-dialing is used the internal numbering must be adapted to the subscribed number series.

The size of the system may have an influence on the number scheme used for internal dialing. It is possible to choose between 2 to 10 digit numbers for extensions. One digit numbers can be used for Hospitality.

After having decided upon the length of the extension numbers, make a plan on how the internal numbering plan should be used:

- Call number to Operator Group and to the individual Operator.
- Call number to PSTN
- Call number to the Private network if applicable
- Call numbers to groups, for example, Group hunting groups
- Number series for Speed dial numbers (Abbreviated Dialing)

4.6.1.3 *Customer Numbering plan*

Customer numbering plan provides an ability to give multiple groups of users in the system their own dialing plan, e.g. multiple groups of users can have the same local dialing plan consisting of the last 2-5 digits of the system numbering plan.

For calls to other user groups, a complete system number must be dialed.

The primary scenario for the feature is when a private network is collapsed into a single system. The users can continue to use the existing dialing plan, e.g. if the network dialing plan was location code based, the users would still dial the short local number and use the location code prefix to dial other locations.

In the system, the users are defined with the long numbers, i.e. including the location code.

Note: For direct in dialling calls from the public network, number conversion must be changed to not just truncate part of the received number but also to prefix the remaining number with the applicable location code.

For new installations where locations would like to use part of the system numbering plan for local calls, it is recommended to use the last digits of the public listed directory number including the area code as system number.

4.6.1.4 *Existing System*

Is there a need to adapt the numbering plan to the numbering used by a previously installed telephone system? The users might want to keep their existing directory numbers.

Make a list of the existing numbers that should be used in the new system.

4.6.1.5 *Trunk Call Discrimination*

It is possible to create a table in the system with public or private network (mostly expensive) destinations, for example, international calls or pay-per-call services, that should be opened or blocked for certain extensions in the system. It is good to create a list of these destinations already in the planning stage.

4.6.2 PRIVATE NETWORK

If the system that should be installed will be part of a private network it is important that the number series is adapted to this existing network.

4.6.2.1 *Open Numbering Plan/Location Code-based Numbering Plan (Uniform Numbering Plan)*

This type of numbering plan is also called **variable length numbering scheme**.

An open numbering plan is where each location (exchange) in the private network needs a unique identifier, a location code, since different exchanges in the network can have extensions with identical directory numbers. The location code is necessary to distinguish two extensions with identical directory numbers but in different exchanges.

4.6.2.2 *Hierarchic Numbering Plan*

If TON (Type Of Number) is possible to convey with the calling/called/connected numbers within the network, it is possible to divide the numbering plan into two different levels, Level 1 Regional and Local private.

4.6.2.3 *Closed Numbering Plan (Coordinated Numbering Plan)*

This type of numbering plan is also called **fixed length numbering scheme**.

A closed numbering plan is used in a private network where there is no conflict between the first 1, 2 or 3 digits in the directory number series. This means that it is not necessary to use a unique identifier (location code) as for an open numbering plan. Any extension in the network is reached by dialling the directory number of the extension, irrespective of in which exchange the calling party is situated.

4.6.2.4

Mixed Numbering Plan

A mix of open numbering plan and closed numbering plan in the same network is called a mixed numbering plan.

4.6.2.5

Common Corporate Numbering Plan

An application where the Routing and Number conversion features may be necessary is the realization of a Common Corporate Numbering Plan. Such a number plan has the characteristic that all corporate extensions are uniquely identified and can be reached from any corporate location by dialing a specific corporate directory number. The exchanges may be connected through tie line, PSTN and/or other networks, in any configuration. The Routing and Number conversion features are used to modify the called or calling number when routing the call through PSTN and other networks.

4.6.3

NUMBER CONVERSION

In a networked scenario or when a call is being forwarded to a user in the public network (PSTN), it might be necessary to make number conversions to adapt the incoming number to the A-number that is to be transferred to PSTN.

For more information on number conversion, see operational directions for *NUMBER CONVERSION AND BEARER CAPABILITY SUBSTITUTION*.

Note: It might not be allowed to send received A-party information to a public or mobile network without concession from the network operator.

Charging and prefixing can be other issues that might have to be considered.

4.7

MEDIA GATEWAY AND NETWORK REQUIREMENTS

4.7.1

GENERAL

MX-ONE 6.x supports media gateways, also remotely located, as described in the paragraphs below. The following limitations and observations must be considered in the MX-ONE 6.x release.

- A server shall only handle one type of media gateway, MGU board or LSU-E
- Remote MX-ONE Media Server is not supported.
- MGU on the same network segment handles shorter latency than MGU on different network segments. This is due to the ARP cache that is not implemented for single network segment.
- MX-ONE supports the possibility to specify which media gateway's resources an operator shall use.
- MX-ONE supports the possibility to set own area code per media gateway
- MX-ONE supports the capability to select outgoing trunk in calling party's own media gateway as first choice for a server.

4.7.2

BANDWIDTH CONSUMPTION FOR A MEDIA GATEWAY

The bandwidth requirement for a media gateway depends strongly on the types of terminals and trunks that are connected. In order to give a rough idea about the

required bandwidth, this paragraph will just give two examples that have been measured in lab environment, using MGU board, with a call intensity of 2 calls per second.

Note: 2 calls/sec is a high calling rate for 1200 users.

- 1) Gateway with ISDN trunks and Digital Extensions
9 Kbytes/s from the server, 4.5 Kbytes to the server
- 2) Gateway with ISDN trunks and SIP extensions
7.5 Kbytes/s from the server, 7.5 Kbytes to the server

Note: This is only the call control bandwidth; the main consumer of bandwidth is the media streams. That bandwidth must be calculated separately and be added to get the total.

4.7.3

SUPPORT FOR MGU IN MIVOICE MX-ONE 6.X

MGU - the MGU board can exist as a gateway, also remotely located, both on the same network segment as the server and on a different network segment.

Single network segment:

Measurements made, with 2 gateways, on the same network segment, show that the round trip latency 100 ms (for the media) is supported at a call rate of 1,2 calls/second. A call rate of 3 calls/second will limit the supported latency to 40 ms round trip (the latency between the 2 gateways cannot exceed 80 ms round trip for the media).

Different network segments:

A call rate of 3 calls/second is supported with a round trip latency of 100 ms, no limitation (except for the max 15 per server) in relation to the number of gateways.

4.7.4

SUPPORT FOR LSU-E IN MIVOICE MX-ONE 6.X

LSU-E, the LSU-E can exist as a media gateway, also remotely located, both on the same network segment as the server and on a different network segment.

Single network segment:

Measurements made, with 2 gateways, on the same network segment as the server, show that the round trip latency 100 ms is supported at a call rate of 1,2 calls/second.

Different network segments:

A call rate of 1,2 calls/second is supported with a round trip latency of 100 ms, no limitation (except for the max 15 per server) on the number of gateways.

5

MIGRATION PLANNING

When planning to migrate from a Telephony Switch to an MX-ONE version 6.x, a number of issues should be addressed, such as hardware and software requirements, traffic requirements and any other differences between the current system and the target system.

For the complete list of requirements, see installation instructions for *MIGRATING MITEL TSW/MD110 TO MIVOICE MX-ONE 6.1*.

6 SITE PLANNING

To run the installation without any problems, it is very important that all requirements for the room are fulfilled. This should be done in proper time, to avoid any delay of the installation. This is valid for power, earthing, lighting and so on.

For the complete list of the requirements, see environmental specification for MIVOICE *MX-ONE SITE PLANNING*.

7

POWER PLANNING

7.1

MIVOICE MX-ONE POWER

The MX-ONE can be powered by the following:

- AC/DC power system, 19" rack mounted power.
- AC/DC power modules, internal open frames, mounted into the 1U and 3U chassis.

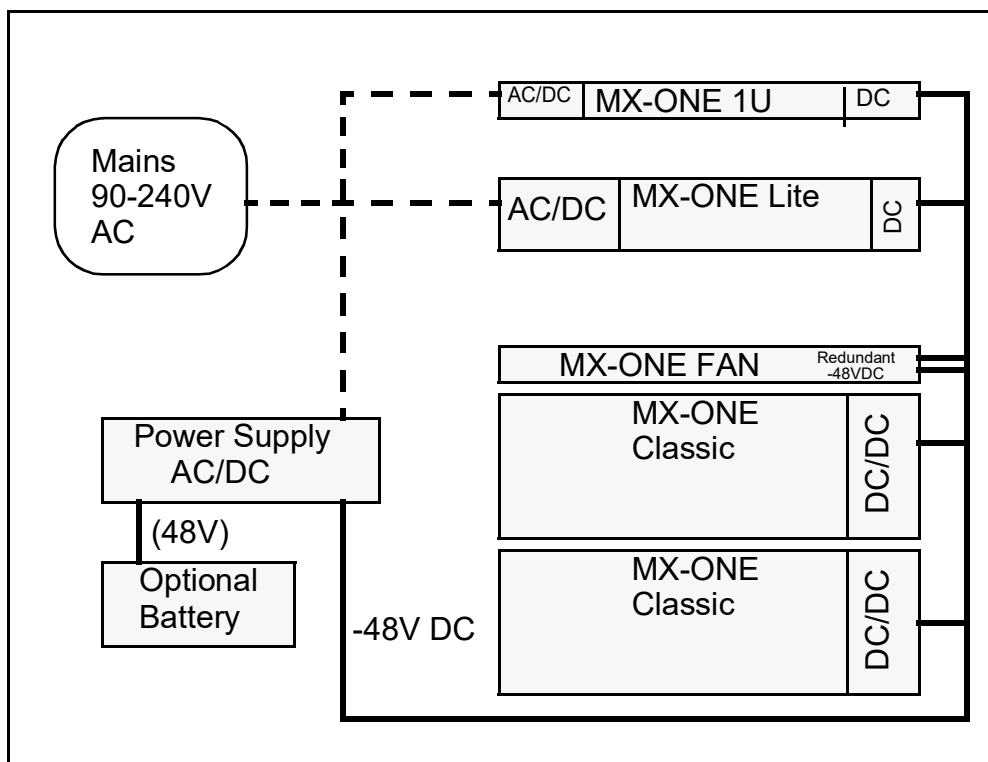


Figure 13: MX-ONE Power Supply system

7.1.1

AC/DC POWER SYSTEM, ASPIRO FROM POWER ONE

The AC/DC power system converts (rectifies) the mains power (90 ~ 240 V AC) to supply -48V DC to the different types of MX-ONE chassis.

The AC/DC power system consists of AC/DC modules (rectifiers), distribution unit and battery connections. It is to be mounted in a 19-inch cabinet.

The ASPIRO power system from is a module based AC/DC system where the AC/DC modules (rectifiers) are "hot swappable", short-circuit proof with microprocessor-based supervision. The Aspiro comes in one version, with optional equipment.

Table 12 Functions in the power system

Function	
Max. number of 800W rectifiers BML351058/1	2, (1600W)
Back-up battery support, see 1U description for details	Yes, 1 string* (battery pack)

Function	
AC redundancy capable	No
Primary controller PCC (needed for correct battery charging)	Optional
Number of fuse breaker outputs	4 (15A)
Number of battery fuse breakers	1 (40A)
Battery cable kit	Optional

* A string is one battery set, four times 12V batteries

7.1.1.1

Power Unit, 800-1600W, product number 51305282



Figure 14: Power Unit (optional equipped with extra 800W rectifier and PCC)

The basic configuration of the power system is with one rectifier (800W), 4 fuse breaker outputs (15A) and one battery fuse breaker (40A). The controller unit, PCC (51305283), is optional and needed when batteries are used or if local/remote supervision is required. See note below.

Note that the cable kit (51305284) for batteries and charging supervision is sold separately.

Two slots are available for rectifiers where one is fitted by default.

Note: It is possible but not recommended to use batteries without the controller. No temperature compensation or settings/status of the batteries can be achieved without the PCC controller.

7.1.2

HIGH AVAILABILITY POWERING

High availability powering can be achieved in several ways.

Redundant AC/DC rectifiers, using at least one more rectifiers than needed to have extra resources when any of the rectifiers is failing.

Battery backup, battery backup is supported. Note that the PCC controller and cable kit is needed when using batteries.

These types of redundancies can also be combined to achieved even higher power availability.

7.1.2.1

Rectifier redundancy

Using at least one more rectifier than the power requirement will give rectifier redundancy. This should normally be combined with the PCC so attention can be distributed when a rectifier fails.

Note that a more than one power system is required if the maximum power consumption is more than 800W.

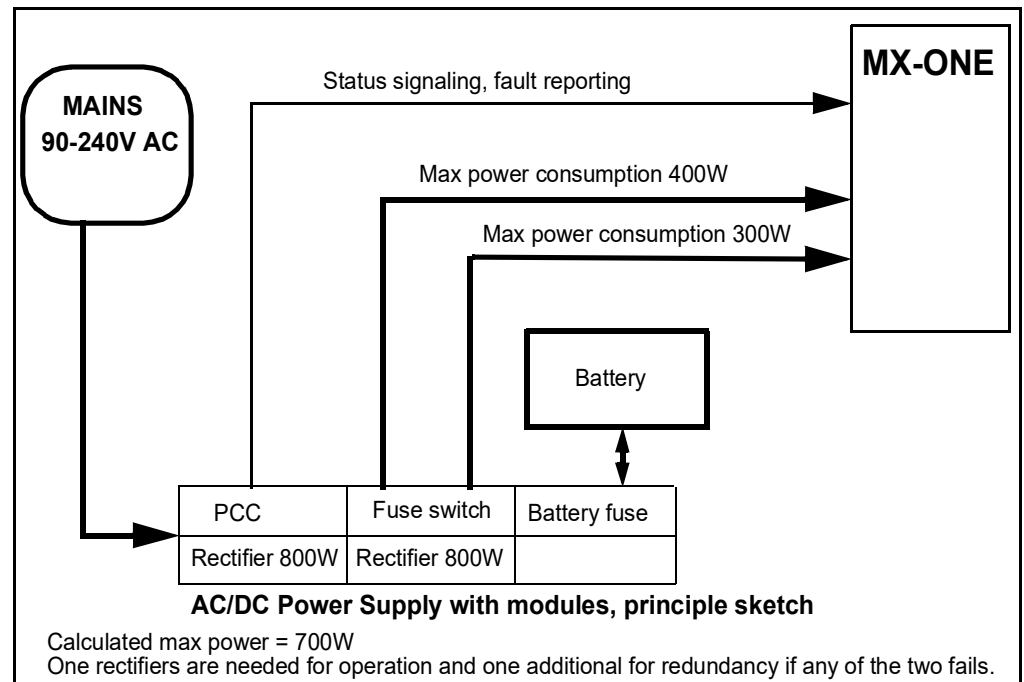


Figure 15: MX-ONE Power Supply with redundant rectifier

7.1.2.2

Battery Backup

The battery backup system can be connected to the AC/DC power system to provide Uninterrupted Power Supply (UPS) if the mains fails.

Batteries can be ordered with two capacities, 31 Ah or 62 Ah. The choice depends on the total power load and the required backup time.

The MX-ONE Classic consumes typically 250W depending on configuration, see 7.4.3 MX-ONE Classic, BFD76140/1 on page 51.

Example: A 62 Ah battery can provide:

-> $62\text{Ah} / (250\text{W} / 48\text{V})\text{A} = 12$ hours of backup time.

This is only a rough calculation and the accurate power planning must be done considering total power consumption of all the equipment in the system that the battery must supply the power to. The battery backup system must be able to supply uninterrupted power during mains failure up to the maximum required backup time.

For details of the power installation, see installation instructions for *INSTALLING MIVOICE MX-ONE HARDWARE* and the Suppliers manual for the power systems.

7.2

AC/DC POWER MODULES

AC/DC power modules mounted inside the 1U and 3U chassis are of the type open frame. The power modules is not monitor able as the AC/DC power system are.

Using of both power modules (in 1U and 3U chassis) and AC/DC power systems like the Aspiro will give redundant power but there is no way to see the actual status. No indication is given if the power modules is failing.

7.3 DC/DC POWER BOARD, ROF1376303

The DC/DC power board is resident in the MX-ONE Classic subracks and distributes the -48 V DC from the AC/DC power system to the backplane.

The DC/DC power board generates the DC voltages, +/- 5V, +/-12V, necessary for the device boards in MX-ONE Classic.

The DC/DC power board also has input for external alarms. See the installation instructions "Installing *MIVOICE MX-ONE, HARDWARE*

Note: A fuse of maximum 15A is required if a third party AC/DC power system is used.

7.4 POWER LOAD CONSIDERATIONS

The power planning must consider the total load per power system and whether a battery backup is required or not.

7.4.1 MX-ONE 1U, 87L00032BAA-A, 1U CHASSIS

The AC/DC power module in MX-ONE 1U can deliver 115W(1) when supplied by 90-240 VAC. 150W can be achieved by using the -48V DC inlet. (2)

Note that, as the circuit boards are mounted horizontally, the internal cooling fan forces the air horizontally (from right to left). It is important not to limit the air flow.

(1) The optional AC/DC power module delivers 125W, 10W is consumed by fans and backplane.

(2) Limited by cooling capabilities.

7.4.2 MX-ONE LITE, 87L00039BAA-A, 3U CHASSIS

The optional AC/DC power module in MX-ONE Lite can deliver 240W* when supplied by 90-240V AC and 300W when supplied by -48V DC.

Note that, as the circuit boards are mounted horizontally, the internal cooling fan forces the air horizontally (from right to left). It is important not to limit the air flow.

Both the optional power module and the -48V inlets are fuse protected.

* The i power supply delivers 250W, 10W is consumed by fans and backplane.

7.4.3 MX-ONE CLASSIC, BFD76140/1

The typical power load of the MX-ONE Classic is approximately 250W, but the actual load highly varies depending on the configuration.

8 APPLICATION PLANNING

8.1 MIVOICE MX-ONE

The applications mentioned below are all optional, and may or may not be used in a particular MX-ONE system. The set of applications that are applicable will also differ between CPE solutions and Cloud solutions.

There are of course also other applications, both from Mitel and from third party, that may require some planning before installation and activation.

8.1.1 INTEGRATION WITH MICROSOFT LYNC SERVER 2013 AND MICROSOFT EXCHANGE SERVER 2013 UNIFIED MESSAGING

It is possible to integrate the MX-ONE with the Microsoft Lync Server 2013, and the Microsoft Exchange Server 2013 Unified Messaging.

For information on the actions needed to integrate with Microsoft Lync Server 2013, see the description for Quick Setup Guide - Integration of Mitel MX-ONE and Microsoft Lync Server 2013.

For information on the actions needed to integrate with Microsoft Exchange Server 2013 Unified Messaging (UM), see the description for Quick Setup Guide - Mitel MX-ONE 6.x Integration with Microsoft Exchange Server 2013 UM.

8.2 MITEL MOBILE CLIENT, MITEL MOBILE CLIENT/MITEL MOBILE CLIENT+

For information on planning of Mitel Mobile Client and Mitel Mobile Client+ deployment, see *Installation and configuration guide for Mitel Mobile Client Controller and Mitel Mobile Client+*, and *Mitel Mobile Client Administration Portal*.

Mitel Mobile Client, single mode, and Mitel Mobile Client+, which is dual mode, are both supported.

8.3 MITEL BLUSTAR FOR PC

The Mitel BluStar for PC client is a soft-clients using SIP signaling, i.e. they must be initiated as SIP extensions in the MX-ONE Service Node. For planning and configuration matters, refer to the *Mitel BluStar documentation portfolio*.

8.4 MITEL COLLABORATION MANAGEMENT, CMG

For planning and for information on how to install Mitel CMG, see *the Mitel BluStar Collaboration Management documentation portfolio*.

8.5 MITEL INATTEND

Mitel InAttend is a SIP based application for attendant functions. It uses SIP trunk connections to the MX-ONE. For planning matters, refer to the *Mitel InAttend documentation portfolio*.

8.6 INCONFERENCE

InConference is an application for conference/multi-party calls. It uses SIP trunk connections to the MX-ONE. For planning and configuration, refer to the *InConference documentation portfolio*.

8.7 MICOLLAB ADVANCED MESSAGING

MiCollab Advanced Messaging is used for Voice Mail and Fax. It uses either SIP extension or SIP trunk connections (with special SIP route profiles). For larger systems, the trunk alternative is recommended. For planning matters, refer to the *Installation & Administrator's guides for Voice Mail and Fax Mail*.

8.8 MICONTRACT CENTER ENTERPRISE

MiContact Center (MiCC) Enterprise is a call center application, which uses CTI groups, extension agent devices and trunk resources in the MX-ONE Service Node, and thus the dimensioning of these resources must be planned.

For information on the requirements that MX-ONE needs to fulfill before integrating with MiContact Center Enterprise, see *Configuring MiVoice MX-ONE for Open Application Server*, delivered with the MiContact Center Enterprise documentation.

8.9 THIRD-PARTY PRODUCT INTEGRATION

8.9.1 CALL ACCOUNTING APPLICATIONS

There are many post-processing applications for Call Information Logging (CIL) (and Quality of Service (QoS) logging) data. For example SofTec Accounting, Metropolis OfficeWatch, and Call Accounting Mate, all third-party products.

It is possible to connect a call accounting application to the MX-ONE. The MX-ONE stores all call records with all fields on the hard disk of the MX-ONE Service Node. Data can be stored in any of the following formats: SQL, XML and CSR. This information can be exported to an external third-party product for post processing of the Call record data for presentation purposes.

9 REMOTE ACCESS

An employee that is working in any (public) location at any given time must be able to access corporate network services. This is also valid for management applications: in particular, it is needed to be able to manage an IT system remotely and hence gain in efficiency and ease of use.

9.1 REMOTE ACCESS SETUP

See corporate guidelines/recommendations on how to set up VPN, security, etc. for remote access.

9.2 REMOTE ACCESS FOR USERS

This section comprises the recommendations as how to use MX-ONE user applications remotely.

9.2.1 SOFT-CLIENTS OR IP TERMINALS

Soft-clients like Mitel BluStar client, MiCollab client, or IP terminals like Mitel 6900/68006700 SIP terminals, could be used in remote access scenarios with a VPN configuration.

It is not recommended to use the remote access clients/terminals without a VPN configuration (for example, with only a firewall) due to their use of several TCP ports that would complicate the firewall configuration.

9.3 REMOTE ACCESS FOR ADMINISTRATORS

This section comprises the recommendations as how to use MX-ONE administrator applications remotely.

9.3.1 MX-ONE SERVICE NODE MANAGER

MX-ONE Service Node Manager (SNM) is a web-based tool and can therefore be used in any remote access scenario, as long as firewalls can be configured according to SNM port configuration, achieved when setting up the system.

9.3.2 MX-ONE PROVISIONING MANAGER

MX-ONE Provisioning Manager (PM) is a web-based tool and can therefore be used in any remote access scenario, as long as firewalls can be configured according to PM's port configuration, achieved when setting up the system.

10

DATA BACKUP

All MX-ONE components (MX-ONE, MX-ONE Provisioning Manager, and MiCollab Advanced Messaging) are equipped with tools to create a backup of the system. Beside the tools, the system administrator needs to implement policies as when to perform data backups and how and where to store them.

The MX-ONE can perform a complete data backup by making use of the MX-ONE Service Node Manager (SNM): relevant files are stored in the file system. The system administrator needs to work out a suitable policy and periodically perform a data backup as well as to store the files in a safe place on a non-volatile media. If multiple MX-ONE Service Nodes are connected together to form a single PBX, it can be chosen to store all the local data backup files on a single Server, from where all files can be handled jointly and stored on some external media storage.

The MiCollab Advanced Messaging (Voice Mail and Unified Messaging functions) provides three methods for backing up data:

- Using the Archive utility to back up the server's databases, configuration settings, mailboxes, announcements, and messages
- Launching the Archive utility automatically after the server's daily maintenance routine
- Manually copying the backup files created during daily maintenance to back up the server's databases

It is recommended to make periodic backups of the server's databases to a location outside the server's platform, to be able to recover the server if there is a problem.

The MiCollab Advanced Messaging Fax server stores most of its data (configuration data, fax mailboxes, and so on) in an SQL database. The Fax server does not include any SQL utilities designed to back up or restore the database. Several SQL backup and maintenance utilities are available commercially and it is strongly recommended to use one of these utilities for creating frequent and regular backup copies of the Fax Server. Beside the content of the SQL database, the fax images should be backed up as well.

For further details about the available backup tools, refer to the specific documentation of the MX-ONE components.

11

VIRTUALIZATION

The MX-ONE Service Node can run on a server which uses virtualization. Certain deployment planning is required regarding which applications are running where, and the performance/capacity of the used server must be sufficient for virtualization and the intended applications. The virtualized environment provides certain redundancy and availability options, that can replace/complement other server redundancy functions.

MX-ONE Service Node virtualized solution runs on top of a VMware infrastructure, vSphere ESXi 5.x hypervisor.

Note: This means that of the ASUs, only ASU-II supports virtualization.

For configuration and planning of virtualization, see the description *MiVoice MX-ONE Service Node Virtualization*.

12

AVAILABILITY AND REDUNDANCY

Since the main function of MX-ONE is to offer telephony services, high availability and redundancy techniques are topics of the utmost importance. This section underlines some aspects related to availability and redundancy that need to be addressed in the planning phase.

12.1

AVAILABILITY

All servers that are part of MX-ONE are based on industrial hardware platforms with extremely high availability standards. All servers are equipped with redundant fans units, and can be provided with additional hard disks in a RAID1 configuration and additional Power Supply Units (PSU). It is strongly recommended to always include the additional hard disk (with RAID 1) and the additional PSU when choosing and configuring the MX-ONE servers.

Note: SW RAID 1 is supported on Mitel ASU and ASU-II.

Particularly important when planning for availability is how to set up the power supply network. Each redundant network requires a separate power supply. For more information about power planning, see 7 Power Planning on page 48.

12.2

REDUNDANCY

The following types of redundancy are supported in MX-ONE:

- Network Redundancy
- Server Redundancy
- HLR Redundancy
- Virtualization with VmWare High Availability and Fault Tolerance

12.2.1

NETWORK REDUNDANCY

To achieve network redundancy, a redundant network infrastructure must be available.

12.2.1.1

Network Redundancy design

Redundant Network can be designed in several different ways, it will depend of the customer requirements, and normally a good network design includes a combination of hardware and software techniques.

Network single point of failure reduction can be achieved using duplication of some hardware and the introduction of some network protocols that can help the availability of the network. Equipment can also be installed in different physical locations.

Techniques like Virtual Local Area Networks (VLAN), Virtual Redundant Route Protocol (VRRP) and Layer 3 Protocols can be used to create a redundant environment that can increase the network availability. When using VRRP different VRID shall be used to avoid that the same MAC address is used by two VLANs.

12.2.1.2

Network redundancy by using Ethernet bonding

By using Ethernet bonding, a switched network with a single subnet can be used for network redundancy.

When using Ethernet bonding, two Ethernet interfaces are aggregated to a logical unit where one interface is active at a time, while the other interface acts as a backup. The two interfaces share the same IP and MAC addresses. If one of the interfaces fail, the other one will continue to serve the operations and the Telephony Servers will be available on the functioning interface.

Ethernet bonding is the only supported method in the MX-ONE Service Node.

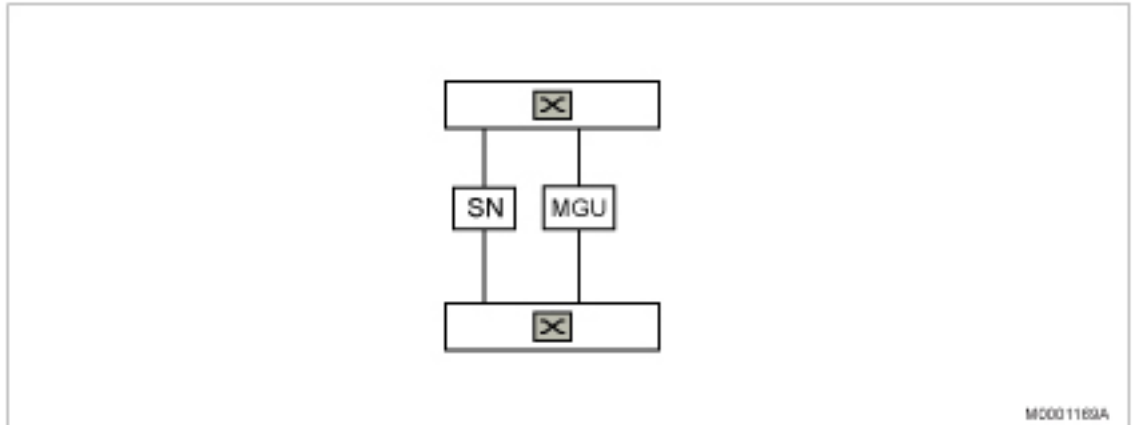


Figure 16: Connection for MGU

Observe that some kind of Spanning Tree Protocol is needed in the network if it contains redundant network paths. It is needed to avoid Ethernet broadcast storms. The performance on network recovery varies depending on type of Spanning Tree Protocol and network configuration. The Rapid Spanning Tree Protocol (RSTP or MSTP) is recommended.

For details see description for *NETWORK REDUNDANCY*.

12.2.1.3

Network Redundancy Considerations

Certain features, like for example operator queue and ACD backup group, can be duplicated and placed in different MX-ONE Service Nodes. This increases the reliability for specific features.

12.2.2

SERVER REDUNDANCY

Server redundancy is achieved by adding one Standby Server per maximum 10 MX-ONE Service Nodes to the network with the ability to take over any failing server. At failover the standby server will take over the identity of the failing server and the control of the media gateway or gateways for the failing Server.

The server redundancy solution in MX-ONE uses the ARP protocol. It sends gratuitous ARP (GARP) messages when a server fails. ARP is used to indicate which hardware address that is connected to a specific IP address. This information is then stored in an ARP table. With GARP, the heartbeat program forcibly updates these ARP tables with new hardware (MAC) addresses when the primary server fails, thereby enabling communication with the backup server.

For details see description for *SERVER REDUNDANCY*.

12.2.3 HLR REDUNDANCY

HLR Redundancy is an alternative to server redundancy where IP extensions (SIP, H.323) can register to a backup HLR server if the main HLR server fails. For more information, see the description for *HOME LOCATION REGISTER REDUNDANCY*.

12.2.4 VIRTUALIZATION WITH VMWARE HIGH AVAILABILITY AND FAULT TOLERANCE

VMware vSphere High Availability allows a cold standby solution, which means in the event that a physical server where the MX-ONE Service Node guest machine is running goes down, a short downtime will occur, due to the fact that MX-ONE Service Node guest machine needs to be initiated in another physical server.

VMware vSphere Fault Tolerance enable a transparent failover solution. This means that in the event that a physical server where a MX-ONE Service Node guest machine is running goes down, no calls will be dropped during the fail-over process and continuity will be maintained.

For more information, see the description for *VIRTUALIZATION*.

13

MIVOICE MX-ONE FUNCTIONALITY USING BLADE SERVERS

13.1

BLADE SERVERS IN MIVOICE MX-ONE

Blade Servers are computer servers with a functionality that does not differ from a standard PC server. The difference lies in that Blade Servers are designed for high density purpose. Each blade (or board) in the Blade System enclosure contains parts that form a separate server. Hardware like power and cooling units are normally shared, and a number of components have been moved outside the server case. This solution makes it possible to enclose more servers per volume compared to standard PC servers, but the setup and configuration of the rack requires more attention.

After installing MX-ONE related software in the HP Blade Server, the server can act as a MX-ONE Service Node.

Note: Blade Servers are not part of the MX-ONE, but a number of setups have been verified to ensure that it is possible to use Blade Servers as a part of a fully functional MX-ONE solution.

14

OPERATION AND MAINTENANCE

Once put into operation, MX-ONE needs to be managed and maintained. This section covers the operation and maintenance aspects that are relevant during the planning phases.

Note: MX-ONE servers can only handle installation of 32-bit Windows 2003 Standard Edition.

14.1

MX-ONE SERVICE NODE MANAGER

The following options are available to configure the MX-ONE:

A Shell for MX-ONE Service Node (MDSH) where access to it can be granted through Secure Shell (SSH). It is therefore necessary to install an SSH client on some host in the network and to make sure that the client can reach the SSH server located on the MX-ONE through the network. Additionally it is necessary to create Linux users on the MX-ONE Service Node entitled to access the node remotely. SSH uses TCP port 22.

The MX-ONE Service Node Manager (SNM) is a Web-based tool which is used to:

- Configure the MX-ONE (for example configure number plans, routes and trunks)
- Back up and restore data on the MX-ONE
- View information on hardware and software revision status for components of the MX-ONE Service Node and Media Gateway
- View security logs, audit trail, and event logs

A Web browser located on a host in the corporate LAN is needed to access the SNM tool. Both HTTP (TCP Port 80) and HTTPS (TCP Port 443) are supported. If HTTPS is used, this needs to be configured.

The following option is available to monitor the MX-ONE: SNMP interface: see 14.6 SNMP on page 62

14.2

MX-ONE PROVISIONING MANAGER

MX-ONE Provisioning Manager (PM) is a web-based tool which is used to manage user and administrator settings.

Most of the functions of PM require relatively little processing capacity and complete quickly. However, some functions, such as listing a large number of extensions can have long response times.

In systems with PM with 4000 extensions or more, we recommend:

- To use a standard server for MX-ONE Service Node (SN) number 1. Or;
- To dedicate a separate MX-ONE server for PM. This does not improve all response times, but does off-load the number 1 MX-ONE server to allow it more capacity for call processing.

14.3 MX-ONE TRAFFIC MANAGER

MX-ONE Traffic Manager (TM) is a web-based tool which is used to manage Traffic Recording data

For details see the *MX-ONE TRAFFIC MANAGER INSTALLATION GUIDE*.

14.4 MITEL COLLABORATION MANAGEMENT, CMG

For more information about Mitel Collaboration Management (CMG), see the installation instruction.

14.5 MICOLLAB ADVANCED MESSAGING

The following options are available to configure MiCollab Advanced Messaging:

Windows GUI: The MiCollab Advanced Messaging Server runs on the Microsoft Windows 2012 Server or 2008 Server OS. A Windows GUI has been implemented to configure a number of settings. To use the GUI, direct access to the server is needed. As an alternative, a remote desktop tool can be used. PcAnywhere is the recommended tool: it needs to be installed on the MiCollab Advanced Messaging Server (pre-installed when delivered) and on a remote host that will be used as a management station. Network connection must exist between the two nodes (TCP port 5631 and UDP port 5632).

The following options are available to monitor MiCollab Advanced Messaging:

- SNMP interface: see 14.6 SNMP on page 62

14.6 SNMP

Each MX-ONE server is equipped with an SNMP agent. Direct integration of these agents with an existing SNMP-based Management framework requires manual configuration of each agent. In particular, parameters such as the IP address of the SNMP manager and the SNMP community must be provided. SNMP messages make use of UDP port 161 and 162.

The SNMP MIB format used in MX-ONE 6.x differs from earlier versions, but is similar to the MX-ONE 5.0 format.

15

IP PROTOCOLS AND PORTS

15.1

STANDARD PROTOCOLS

Apart from standard TCP/IP network protocols like ICMP, TCP, the following protocols are used by the MX-ONE.

The table of ports and protocols is not complete or definitive, but covers voice connectivity. Additional optional applications or service may add other ports and protocols. See the particular application's documentation for details.

The column marked Firewall indicates that the port should be open if a firewall and NAT is used. Y = port should be open, (Y) = port should be open if the involved entities are at different sides of the firewall.

Table 13 Standard Services

Service	Direction	Usage	Ports& Protocols	Fire wall	Notes
ARP	MX-ONE Server -> Network	Gratuitous ARP used by standby-server to update IP address ownership.	-		Address Resolution Protocol (for IPv4), defined by RFC 826
CLVM-CFG	MX-ONE Server -> MX-ONE Server	For internal use. Blocked for incoming connections.	TCP/UDP 1476		
CSTA Phase III (without TLS)	XML client <-> MX-ONE Server or Web Service client <-> MX-ONE Server	Third party call control via the CSTA phase 3 interface	Configurable. Defaults are TCP 8080/80 for Web Service based client and TCP port 8882 for XML based client. TCP port 5062 for TR87.	Y	A standard version of the CSTA protocol is used between the MX-ONE XML or Web Service application suite for third party call control. <u>ASN.1 is no longer supported.</u> TR87 = CSTA III transported on SIP. Both IPv4 and IPv6 are supported.
CSTA Phase III (with TLS)	XML client <-> MX-ONE Server or Web Service client <-> MX-ONE Server	Third party call control via the CSTA phase 3 interface	Configurable. Defaults are TCP 8080/80 for Web Service based client and TCP port 8883 for XML based client.	Y	A standard version of the CSTA protocol is used between the MX-ONE XML or Web Service application suite for third party call control. <u>ASN.1 is no longer supported.</u> TR87 = CSTA III transported on SIP. Both IPv4 and IPv6 are supported. Web Service only supports TLS 1.0.
Daytime	MX-ONE Server -> MX-ONE Server	For internal use. Blocked for incoming connections.	TCP/UDP 13		

Service	Direction	Usage	Ports& Protocols	Fire wall	Notes
DHCP	IP phones -> DHCP Server	To provide the IP address to the telephone, but also the IP address to the HTTP server that holds the configuration data and the firmware files used in the terminals. (Client Side)	-	Y	
DNS	SIP phones -> DNS Server SIP clients -> DNS Server MX-ONE Server -> DNS Server	To find a registrar and zone information	TCP/UDP 53	(Y)	The SRVREC file is stored in the DNS server. MX-ONE: fetching zone information from master DNS
H.225.0 RAS	MX-ONE <-> VoIP Clients	Registration and Admission to make/receive calls. To send Status and Quality Reports to the MX-ONE. IPv4 is supported.	UDP 1718 (GK discovery) UDP 1719 TCP/TLS 3727	Y	H.225.0 RAS is not used in the IP trunking service.
H.225.0 Q.931	All H.323 based devices <-> MX-ONE Server	To set up and tear down different media sessions (voice calls) between all H.323-based communication devices. It is also used in IP trunking service to transfer QSIG data. IPv4 is supported.	TCP 1720 (IP trunk, listening mode) TCP 1722 (IP extension, listening mode) TCP 32768-61000 (gateway client mode)*1 TCP 1024-5000 (IP extension client mode)*2 TCP/TLS 1300 TLS 1300 (IP extension, listening mode). TLS 1301 (IP trunk, listening mode).	Y	*1) The range is defined by the operating system (Linux) and can be changed by command. *2) The range is defined by the operating system (VxWorks).
H.245 Media	All H.323 based devices/end-points <-> MX-ONE MGW	To exchange data about possible media configurations for a particular call. IPv4 is supported.	Dynamically allocated TCP 17002-19492 (IP extension, listening mode) TCP 24002-26492 (IP trunk, listening mode) TCP 32768-61000 (gateway, client mode)*1 TCP 1390-1396 (IP phone)	Y	*1) The range is defined by the operating system (Linux) and can be changed by command.

Service	Direction	Usage	Ports& Protocols	Fire wall	Notes
HTTP / HTTPS	IP phones MX-ONE Service Node Manager <-> IP phones MX-ONE Service Node Manager <-> Media Gateway Unit	HTTP is used to upload the configuration data and the firmware files to IP phones and IP clients. HTTPS is used for Web-based management. (Client and Server side).	80 443	Y	The ports used by the Web interface of the MX-ONE Service Node Manager can be configured to be different from the standard ones for this kind of services (80/443). This has to be considered when configuring the application and the network devices. In the latter case, NAT-boxes, firewalls and other similar devices must be configured to allow traffic flowing through the configured ports.
HTTPS	Mitel Mobile Client -> MX-ONE Service Node Manager	Mitel Mobile Client signals it's extension number and called B-number to MX-ONE Service Node Manager.	9443	Y	
ICMPv6	MX-ONE Server -> MX-ONE Server	Used by SLAAC to obtain IPv6 addresses for the servers. Also used by standby-server to update IP address ownership. Multicast addresses: FF02::1 and FF02::2	-		Internet Control Message Protocol version 6, according to RFC 4443. Not used for auto-configuration. Used initially by MX-ONE to find network addresses, but then turned off.
Kerberos	MX-ONE Server -> MX-ONE Server	For internal use. Blocked for incoming connections.	TCP/UDP 4444		
LDAP	MX-ONE Server <-> LDAP	Communication between servers.	TCP/UDP 389		
LDAPSSL	MX-ONE Server <-> LDAP	Communication between servers.	TCP/UDP 636		LDAP protocol over TLS/SSL
MSCML	MX-ONE Server -> MX-ONE Media Server				See SIP.
NDP	MX-ONE Server -> Network	Used by SLAAC to obtain IPv6 addresses for the servers.	-	Y	Neighbor Discovery Protocol (replaces and enhances ARP).
NTP	MX-ONE Server <-> Network	To update the system time in a controlled manner. (Client side)	TCP/UDP 123	Y	
Postgres	MX-ONE Server -> MX-ONE Server	For internal use. Blocked for incoming connections.	TCP/UDP 5432		
Rpcbind	MX-ONE Server -> MX-ONE Server	For internal use. Blocked for incoming connections.	TCP/UDP 111		Same as Sun rpc

Service	Direction	Usage	Ports& Protocols	Fire wall	Notes
RTP/RTCP	MX-ONE Media Gateways <-> VoIP clients	For media transmission. (Both Client and Server side)	UDP Define a suitable port range for security and capacity reasons for sending rtp/rtcp packets over a network with proxies and firewalls.	Y	See the <i>media_gateway_interface</i> command. Minimum 200 consecutive media ports are required, and they must of course not collide with ports used by other applications. The number of media ports shall be twice the number of RTP resources supported by the gateway type.
RTP/RTCP	VoIP Client -> VoIP Client VoIP Client <-> MX-ONE Media Gateway MX-ONE Server -> SIP phones	For media transmission. (Both Client and Server side)	UDP 16986-17012 (RTP, IP phones) UDP 16987-17013 (RTCP, IP phones)	Y	
RTP/RTCP	MX-ONE Server -> SIP phones	Protocol for media streaming on idle SIP terminal.	UDP 60000	Y	Only Mitel SIP 6800 and later phones models. Default is port 60000, but can be configured. See <i>streaming_data</i> command.
SIP	MX-ONE Server -> VoIP clients/phones	Signaling protocol for Internet conferencing, telephony, presence, CSTA via SIP (TR87), events notification and instant messaging. Both IPv4 and IPv6 are supported.	TCP/UDP 5060 TCP/TLS 5061 TR87: 5062	Y	See also CSTA Ph. III regarding TR87 Regarding XML, see Proprietary protocols.
SIP	SIP phones/clients <-> MX-ONE Server MX-ONE SIP trunks <-> Network	Signaling protocol for Internet conferencing, telephony, presence, CSTA via SIP (TR87), events notification and instant messaging. Both IPv4 and IPv6 can be supported.	TCP/UDP 5060 TCP/TLS 5061 TR87:5062	Y	See also CSTA Ph. III regarding TR87 Regarding XML, see Proprietary protocols.
SIP (MSCML)	MX-ONE Server <-> MX-ONE Media Server	Signaling protocol for Media Streaming Control Markup Language and Protocol. Used for media streaming functions via Media Server. Both IPv4 and IPv6 are supported.	UDP 5090	Y	See RFC 5022 for MSCML. See the <i>media_server</i> command for details.
SMTP	MX-ONE Server -> Network	Feature based license usage reporting	TCP 25	Y	

Service	Direction	Usage	Ports& Protocols	Fire wall	Notes
SNMP	MX-ONE Server -> Network Mitel MiCollab Advanced Messaging -> Network	For exchange of management information between network devices	UDP 161	Y	
SNMP Trap	MX-ONE Server -> Network Mitel MiCollab Advanced Messaging -> Network	For exchange of management information between network devices	UDP 162	Y	
SSH	Network -> MX-ONE Server MX-ONE Server -> MX-ONE Server	Can be used to run a remote session on a computer, over a network and perform common management operations.	TCP 22	Y	
TFTP	-> MX-ONE Server	For internal use. Used to update firmware on LSU.	TCP/UDP 69		
Vat	MX-ONE Server ->	For internal use. Blocked for incoming connections.	TCP/UDP 3465		
WAP	MX-ONE Server -> VoIP Client/phone	For display feature services and state information handling. (Server Side)	UDP 9200 ESSP	Y	For H.323 and DECT extensions.

15.2

PROPRIETARY PROTOCOLS

The system makes use of a number of proprietary signaling interfaces, most of them optional. Each of the interfaces allocates a user defined communication port during initiation of the particular service.

Table 14 Proprietary Services

Service	Direction	Usage	Ports& Protocols	Fire wall	Notes
ACS-AppServ	MX-ONE Service Node Manager <-> MX-ONE Server	To communicate with the MX-ONE Service Node via a proxy using JNI (Java Native Interface).	TCP 2000		
AMP	MX-ONE Server -> MX-ONE SNM/PM	For internal use.	TCP 3456		
CFCOPY	MX-ONE Server -> MX-ONE Server	Copy of reload data from active to standby common function.	SCTP 5679		
ConfigServer	MX-ONE server <-> SIP extensions	SIP, Visitor Deskphone (hotdesklogin). Fetch and upload of user configuration files	Methods: GET,POST HTTP:22225 HTTPS:22226	Y	The ConfigServer port is only for SIP extension family Mitel 6800. Set in ip_telephony.conf

Service	Direction	Usage	Ports & Protocols	Fire wall	Notes
CSTA based protocol	MX-ONE Server <-> Mitel Collaboration Management, CMG.	Third party call control	TCP 2599	(Y)	A proprietary version of the CSTA protocol (TCP 2599) is used between the MX-ONE and Mitel Collaboration Management, CMG for third party call control. Regarding TSAPI, see OAS/AL documentation.
DLS, Deployment Service (for Unify phones)	MX-ONE Server -> Unify SIP phones	SIP phone configuration, for Unify (former Siemens) telephones	Configurable Default: Not set Recommended: HTTPS:22224	Y	The DLS port is only for Unify SIP extensions. Requires installed certificate and DLS license.
GICI over IP	MX-ONE Server <-> Mitel Collaboration Management, CMG. (Client Side)	GICI services, like MWI, Message Diversion and VM	TCP 998	(Y)	
HW Signaling	MX-ONE Server <-> Media Resource Gateways	Traditional signaling (LSU) O&M functions	TCP 2816 TCP 2817	(Y)	
HW Signaling, IPLU	MX-ONE Server <-> Media Resource Gateway, IPLU	Tracing functions for IPLU.	TCP 9000 TCP 9001	(Y)	For DSP application tracing.
HW Signaling, IPLU	MX-ONE Server <-> Media Resource Gateway, IPLU	Tracing and trouble-shooting functions for IPLU.	TCP 20-21 TCP 23 TCP 1500	(Y)	20-21 for log file collection using ftp. 23 for DSP internal tracing on Mindspeed execution, using telnet. 1500 for trouble-shooting commands.
Inter-Server Connection	MX-ONE Server -> MX-ONE Server	Inter-server signaling.	SCTP 5678		
MiCollab User Provisioning	MX-ONE PM>MiCollab Server	Application Programming Interfaces, internal use	TCP ports 10255, 10256, 10257, 10258, 10259, and 10260		The ports listed are on MiCollab Server
MX-ONE Service Node - Media Gateway signaling	MX-ONE Server <-> Media Gateway Unit	Used for device board, media and O&M control signalling.	SCTP 2816-2818	(Y)	
PAS, CSTA3 based protocol	MX-ONE Server -> Network (other server)	Telefonica PAS gateway	Configurable Default: 80	Y	PAS gateway is at the network operator
Server Redundancy	MX-ONE Server -> MX-ONE Server	Signaling protocol for Cluster Application, when Server Redundancy is used.	UDP 4466	(Y)	

Service	Direction	Usage	Ports & Protocols	Fire wall	Notes
VSI	Mitel Collaboration Management, CMG <-> Mitel MiCollab Advanced Messaging	Voice System Interface used by MiCollab Advanced Messaging and CMG	TCP 3001	(Y)	
XML	MX-ONE Server -> SIP extension	SIP extension key and display control interface	XML, HTTP:22222 HTTPS:22223	Y	The XML port is only for SIP extension families 6900/6800/6700

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CHECKLIST

This section gives a summary of all the required planning activities described in the previous sections. It can be used as a checklist when planning how to install MX-ONE and integrate it with the existing IS/IT infrastructure.

- 1) Is Site Planning completed regarding the following:
 - Required space
 - Power supply
 - Cooling
- 2) Has the estimated downtime been taken into account if migrating from an MD110 to an MX-ONE system?
- 3) Are the following traffic statistics collected:
 - Maximum call attempts during busy hours
 - Maximum number of users
- 4) If an SLA with a network operator exists, have the following issues been addressed in the agreement?
 - Maximum Delay
 - Packet Loss
 - Availability
 - Required Bandwidth
- 5) Has a network assessment been performed in regard to the following:
 - Switched LAN
 - Filtration
 - Firewalls
 - VLAN
 - Used IP protocols (IPv4, IPv6, or both?)
- 6) Has a delay analysis been performed?
 - Maximum delay is 80 ms
 - Maximum packet loss is 2%
 - Jitter must not exceed 20 ms
- 7) Have the required bandwidth needs been analyzed, including identification of any critical links regarding:
 - WAN
 - Inter-MX-ONE Service Node traffic
- 8) Is there a valid number plan in place?
- 9) Have the security requirements been fulfilled? See the user guide for *SECURITY GUIDELINES*.
- 10) Has the network environment been planned in regards to the following:
 - DNS (IPv4)

- NDP (IPv6)
- DHCP (IPv4)
- ICMPv6
- Multicast

11) Is there a plan for how to set up user and system administration in regard to:

- Remote access

12) Is any type of redundancy to be used?

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DEFAULT AND GUIDING VALUES

Symbol	Meaning
λ	Busy Hour Call Attempts, calls per hour
α	Mean Traffic per access type, normal traffic, (erlang)
A	Total traffic to a device, (erlang)
A_h	Total traffic during high load
A_n	Total traffic per trunk in a group
B_n	Probability of blocking, normal load
B_h	Probability of blocking, high load
$E_{1,n}(A)$	Probability of blocking (Referred to as B in this document)
$E_{2,n}(A)$	Probability of waiting
h	Mean holding time for a call, (seconds)
n	Number of devices
$P(w>t)$	Probability of waiting time w exceeding t seconds
R_h	Ratio Factor for Peak margin, if B_h is not given

17.1

DEFAULT VALUES FOR MIVoice MX-ONE

For MX-ONE, the following values apply:

Symbol	Meaning
α	0.2 erlang /terminal (0.1 internal + 0.1 external)
B_n	0.01
R_h	1.2
h	90 seconds ({60 seconds internal + 120 seconds external} /2)
A_n	0.5 erlang per trunk/ loss resource for groups = 10 channels
A_n	0.6 erlang per trunk/ loss resource for groups 11 - 20 channels
A_n	0.7 erlang per trunk/ loss resource for groups 21 - 35 channels
A_n	0.8 erlang per trunk/ loss resource for groups > 35 channels
n	1-500 devices
A_h	$n \times R_h \times \alpha = h \times \lambda / 3600$

Example: How many trunk lines are needed for 250 IP extensions?

$A = 250 \times 1.2 \times 0.1 = 30$ Erlang, 0 Erlang at $B_h = 0.01 \Rightarrow 40$ trunks are needed.