

MiVoice MX-ONE

Media Gateway Unit, MGU - Description

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Introduction

This document describes the functions, performance and limitations in the MX-ONE Media Gateway Unit (MGU).

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

The key features of MGU includes:

- **Device Board Interface**
MGU intermediates all communication between device boards in the 3U/7U chassis and the MX-ONE Service Node.
- **Digital trunks**
MGU provides layer 1 and layer 2 for E1/T1.
- **VoIP**
MGU provides RTP/SRTP including DTMF detection, DTMF relay and facsimile tones over RTP. The VoIP channel also includes configurable echo canceler.
- **Fax relay T.38**
MGU provides relaying T.30 facsimiles (G3 fax) to/from Internet Aware Faxes or Gateways using T.38 protocol.
- **Keycode Receiver**
MGU provides Keycode Receivers (DTMF and MFC receivers), intended for mobile extensions (DTMF) and CAS E1 trunks (MFC).
- **Keycode Sender**
MGU provides Keycode Senders (DTMF and MFC senders), intended for mobile extensions (DTMF) and CAS E1 trunks (MFC).
- **Tone Sender**
MGU provides Tone Senders for call progress tones, e.g. dial-tone, according to market specifications.
- **Multi Party**
MGU provides Multi Party resources for e.g. conference and intrusion call cases.
- **Recorded Voice Announcements**
MGU provides play out of pre-recorded, locally stored, media files over TDM switch.
- **TDM switch**
MGU provides a non-blocking TDM switch with attenuation support for interconnection of circuit switched media.
- **Network Redundancy**
MGU supports redundant networks.
- **External Alarms**
MGU supports inputs in backplane for external alarms.

Scope

This document provides a description of the MGU board, provided functions, their performance and limitations. The document does not cover details of end-user and administrator commands, etc. provided by MX-ONE Service Node and/or MX-ONE Managers for initiating or using these functions.

Glossary

For a complete list of abbreviations and a glossary, see the description for *Acronyms, Abbreviations, and Glossary*.

AES

Advanced Encryption Standard

AH

Authentication Header

CAS

Channel Associated Signaling (for E1 trunk interface).

CNG

Comfort Noise Generation. Used to generate background noise when no RTP packets are received or when NLP is engaged.

DBS

Device Board Server. This is a MGU subsystem, running in a linux process for device board communication, and also implementing virtual device boards (ISDN).

ESP

Encapsulating Security Payload

IKE

Internet Key Exchange

ISAKMP

Internet Security Association and Key Management Protocol

IPsec

Internet Protocol Security

LFO

Link Fail Over. Redundancy with switched network.

NLP

Non-Linear Processor. Removes the residual echo that the linear echo canceler couldn't remove. It is idle at double-talk.

DFE

Dual-Filter Echo Canceler. This is an optional echo canceler in MGU that improves echo canceling significantly, but also decreases capacity.

DP

Device Processor. This is the main control and supervision processor, running a linux operating system, on MGU board.

DSP

Digital Signal Processor.

MCA

Media Control Application. This is a MGU subsystem, running in a linux process for Media related tasks (VoIP and TDM switch management).

MSP

Media Stream Processor. System-on-Chip module with DSP capabilities for voice and media processing.

OMA

Operation & Maintenance Application. This is a MGU subsystem running in a linux process for O&M related tasks.

PCM

Pulse Code Modulation. Digital representation of analog signals in e.g. circuit switched (TDM) systems. In TDM systems, usually 8000 Hz sampling rate and A-law or mu-law encoding is used.

PRI

Primary Rate Interface (2048 kbit/s E1, or 1544 kbit/s T1).

PTS

Proceed To Send.

SA

Security Association

SLIP

A standardized procedure to take care of different clock rates between two digital systems by either skipping or repeating a frame of data

TDM

Time Division Multiplexing. A way to transfer several channels, containing PCM samples (timeslots), on a single wire.

VAD

Voice Activity Detect. Used to stop RTP packet encoding and transmission during silence periods in received PCM stream, resulting in reduced DSP and network load.

Virtual Board

A native application on MGU that simulates a legacy physical ("real") board. From a management point of view it is configured and behaves like a real board.

Virtual Magazine

The virtual magazine is the equipment range in MGU that holds the virtual boards and the MSP resources that are mapped to equipment numbers.

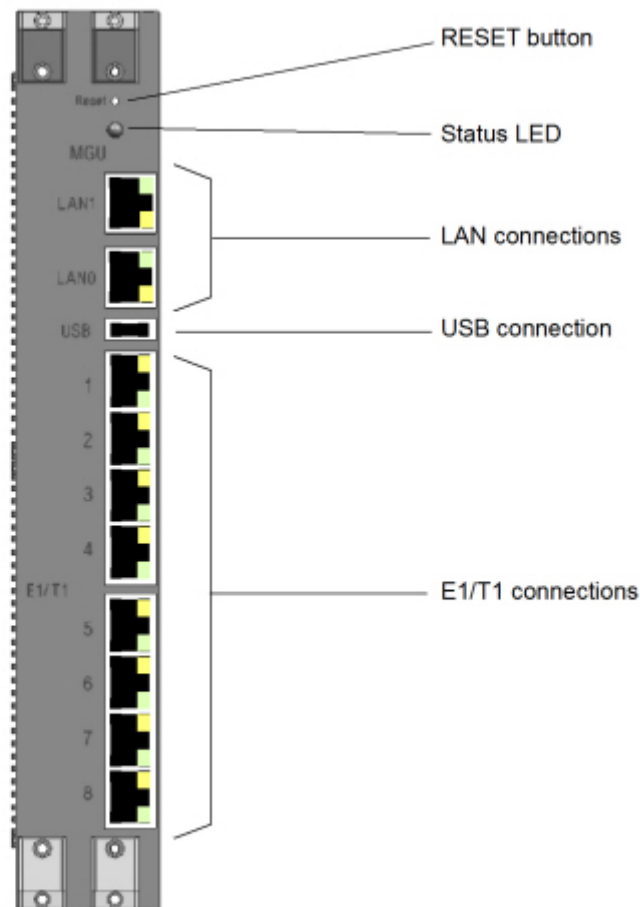
Board Description

This chapter gives a high-level description of the Media Gateway Unit (MGU) board.

Board Layout and Front Connectors

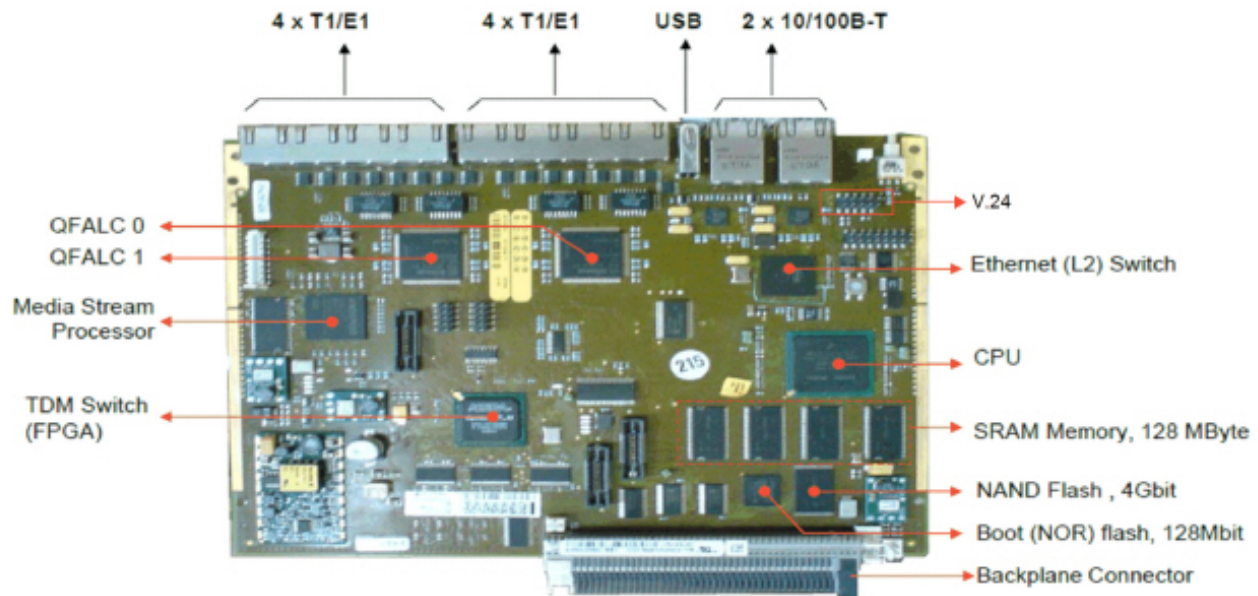
The following figure shows the MGU front and the external connectors in the front. For further description of these, refer to section [Interfaces](#).

Figure 2.1: MGU front and connectors



The following figure shows interfaces and key components on PCB.

Figure 2.2: MGU board layout



Architecture

The MGU hardware architecture and its external interfaces (connectors) is outlined in figure below. The board has connectors in the front and to the backplane. The back-plane is mainly the interface to other device boards, but also includes some alarm signals and power to the board.

Interfaces

- **LAN0 and LAN1**
Primary (LAN0) and Secondary (LAN1) LAN ports. These two ports provides support to connect to redundant networks. See [IP Network Redundancy and Security](#) for more information how to connect these.
- **Device Board Interface** (Backplane Interface)
This is the interface towards all device boards. Up to 16 device boards can be accessed. It includes TDM buses (including frame sync input and output) and HDLC/UART signaling buses to all device boards.
- **E1/T1**
8 Primary Rate Interface (PRI) for ISDN trunks, CAS trunks and CAS Extensions.
- **Visual Indicators** (front LED)

The LED shows the operating status of the MGU board. There are also indicators on the LAN and E1/T1 connectors, see further [Visual Indications](#).

- **Reset button**

The Reset button is connected to HW reset of the Device Processor (DP) and will immediately restart the board.

- **USB**

Management and service interface (linux console) which support USB to serial (V.24) bridge (USB serial dongle). A TSR 899 135/1 cable can be connected from this interface to for example a PC with terminal program. The terminal program shall be configured for 9600 baud, no parity, 1 stop bit to connect to this interface. Other USB serial dongles that uses a PL2303 chip might work as well.

NOTE: All management that can be done from the USB interface can also be done through SSH login.

- **V.24**

The V.24 interface located on the PCB is mainly intended for debug purposes, but can be used as a “fall back” Management interface if USB access is not possible. Same terminal configuration as for USB applies. A TSR 43 297/1000 cable can be connected to this interface.

Key Components

TDM Switch

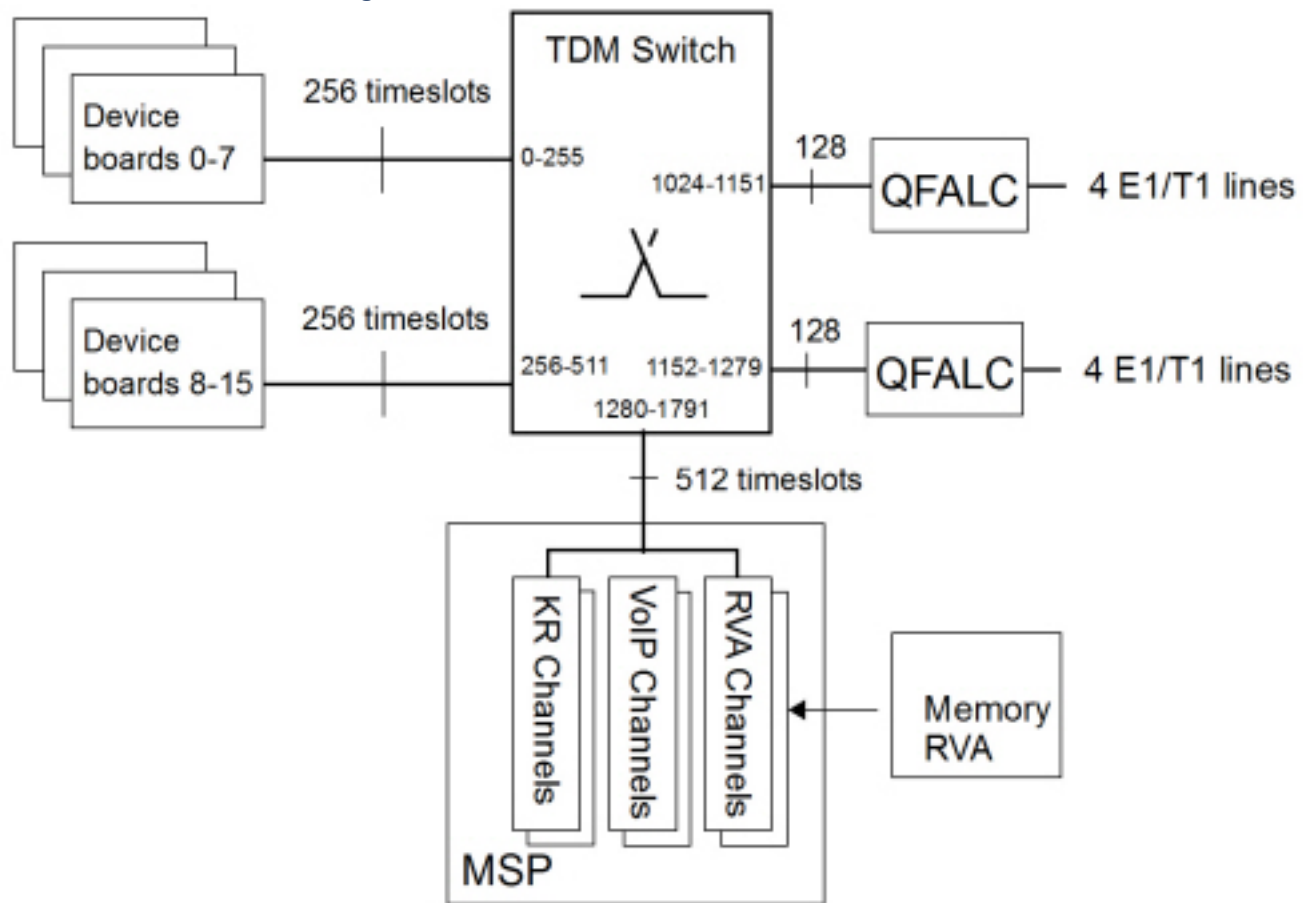
The TDM (Time-Division Multiplexing) switch on MGU has a very central function, since all media interconnections between trunks, extensions, and auxiliary functions are made through this switch. An exception to this is when media is setup between two IP extensions where media can be setup direct between these extensions on the LAN. However, there are also such cases when IP extensions are forced connected via the MGU and thus through the TDM switch as well.

The TDM switch is a non-blocking Time-Space-Time switch handling cross connections for up to 2048 64kbps timeslots. Of these, only 1280 are connected and used, as shown in figure 3 below.

The switch provides a multi-cast feature (also known as sunfan) to connect one timeslot to many other (there is no restrictions how many).

Figure 3 below shows the TDM switch and interconnection of TDM devices on MGU. The figure shows the physical (HW) timeslot numbers. These numbers can be mapped to logical timeslot numbers (multiple numbers) and EQU (Equipment) positions as shown in table See [TDM Switch Timeslots to Equipment/Resource mapping](#).

Figure 3 below shows the TDM switch and interconnection of TDM devices on MGU. The figure shows the physical (HW) timeslot numbers. These numbers can be mapped to logical timeslot numbers (multiple numbers) and EQU (Equipment) positions as shown in table [TDM Switch Timeslots to Equipment/Resource mapping](#).

Figure 2.3: TDM switch and interconnection to TDM devices

For each device connected to TDM switch, the number of 64kbps timeslots is stated, and the physical timeslot number (0-2047) in the switch (note: not all timeslots are used, see table [TDM Switch Timeslots to Equipment/Resource mapping](#)).

The TDM switch provide functionality to select attenuation level on each connection individually. Attenuation values are pre-configured at MGU startup according to selected market. See market characteristics for attenuation levels used per market.

Table 2.1: TDM Switch Timeslots to Equipment/Resource mapping (Sheet 1 of 2)

TDM switch range	Equipment / Resource type		Multiple Number range (decimal / hex)		EQU range *) (within Media GW) **)
0..511		3U and 7U	0..31	000..01F	0-00-0 ..0-00-31
			32..63	020..03F	0-10-0 .. 0-10-31
	Device Boards (16 positions)		64..95	040..05F	0-20-0 .. 0-20-31
			96..127	060..07F	0-30-0 .. 0-30-31
		3U and 7U	128..159	080..09F	0-40-0 .. 0-40-31
			7U only	160..191	0A0..0BF
		192..223		0C0..0DF	0-60-0 .. 0-60-31
		224..255		0E0..0FF	0-70-0 .. 0-70-31
		256..287		100..11F	1-00-0 .. 1-00-31
		288..319		120..13F	1-10-0 .. 1-10-31
		320..351		140..15F	1-20-0 .. 1-20-31
		352..383		160..17F	1-30-0 .. 1-30-31
		384..415		180..19F	1-40-0 .. 1-40-31
		416..447		1A0..1BF	1-50-0 .. 1-50-31
		448..479		1C0..1DF	1-60-0 .. 1-60-31
		480..511		1E0..1FF	1-70-0 .. 1-70-31
512..1023	Unused				
1024..1279	E1/T1 Links (8 virtual boards)	Link 1	512..543	200..21F	2-00-0 .. 2-00-31
		Link 2	544..575	220..23F	2-10-0 .. 2-20-31
		Link 3	576..607	240..25F	2-20-0 .. 2-30-31
		Link 4	608..639	260..27F	2-30-0 .. 2-40-31
		Link 5	640..671	280..29F	2-40-0 .. 2-50-31
		Link 6	672..703	2A0..2BF	2-50-0 .. 2-60-31
		Link 7	704..735	2C0..2DF	2-60-0 .. 2-60-31
		Link 8	736..767	2E0..2FF	2-70-0 .. 2-60-31

Table 2.1: TDM Switch Timeslots to Equipment/Resource mapping (Continued) (Sheet 2 of 2)

TDM switch range	Equipment / Resource type	Multiple Number range (decimal / hex)		EQU range *) (within Media GW) **)
1280..1791	Dynamic Resources in MSP (for example, VoIP channels and Keycode Receivers)	768..1279	300..4FF	3-00-0 .. 3-70-31 4-00-0 .. 4-70-31
1780..2047	Unused			

*) The Media GW number has been omitted from the EQU numbers in this table. For example, if resource is in Media GW 1A, 1A- shall be added as a prefix to the EQUs listed here.

**) As of MX-ONE 5.0 SP3, the following TMU functionality has been moved to the MGU software. Therefore, no TMU board is needed in a MGU based MGW chassis for a standard MX-ONE installation with IP/TDM users assuming that the Mitel InAttend operator is used:

- Extension conference
- Extension Intrusion
- DTMF send/receive
- Tone sending

The following functionality is not supported by the MGU based TMU software.

- Native MX-ONE operator
- Dial tone detection (use a Proceed to send, PTS timer instead)
- Specific market tones (for example, morse)
- Analog MoH input, for example, no live announcement for Emergency notification.

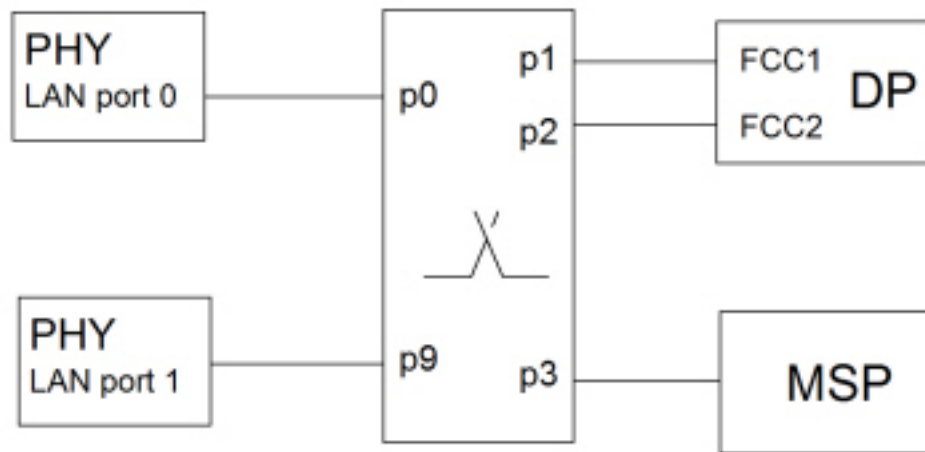
In the case that the above functionality is needed, then a TMU board must be present with the MGU based MGW chassis.

Ethernet (Layer 2) Switch

There are two standard 10/100 Base-T LAN connection on the MGU board marked LAN 0 and LAN 1.

The Ethernet packets are via a layer 2 switch routed to the Device Processor (DP) or to the Media Stream Processor (MSP). See figure below.

Figure 2.4: Ethernet switch



The switch is configured in different ways depending on redundancy scenario (see [IP Network Redundancy and Security](#)). As seen in figure above, the MSP provides only one Ethernet interface which is re-programming to emulate two Ethernet MACs (Subnet Redundancy).

Packets on the LAN can be of two kinds. Non-RTP packets, for signaling, are routed to the DP. RTP packets carrying media (VoIP), are routed to the DSP.

Media Stream Processor (MSP)

The M82710 is a System-on-chip for VoIP telephony applications.

The two main blocks are the Media Stream Processor (MSP) and the Control and Signaling Processor (CSP).

The MSP is responsible for all delay sensitive media processing within the device, and in this case contains two main elements high performance voice band signaling processors (DSPs) and a RISC network processor (ARM).

The CSP is responsible for both the control of the MSP and for running all the applications like IP signaling stacks and management. The CSP is also a RISC network processor (ARM).

The MSP is a common resource pool shared by all DSP-related functions provided by MGU. Examples of functions are VoIP, T.38 and DTMF receivers. Depending on function used and configuration settings, the load on MSP differs (the load from a particular function might even vary over time) and thus the maximum density in MSP varies depends on usage. See [Capacity and Limitations](#) that states MSP density for different functions and a few configurations.

The processor load is continuously supervised and reported to MX-ONE Service Node to indicate high-load conditions (high load conditions will be logged in MGU's syslog as well). High-load condition is also checked in run-time when orders to activate a new DSP function is received from MX-ONE Service Node. If too high load at that time, the order is rejected with information about the cause.

NOTE: The syslog is stored in the file `/var/log/syslog` in MGU's file system.

TDM Synchronization Unit

A clock circuit is used to generate a system clock. The system clock is distributed to internal (on board) components and external resource boards in a magazine where applicable.

The clock source is user defined and is one of the following:

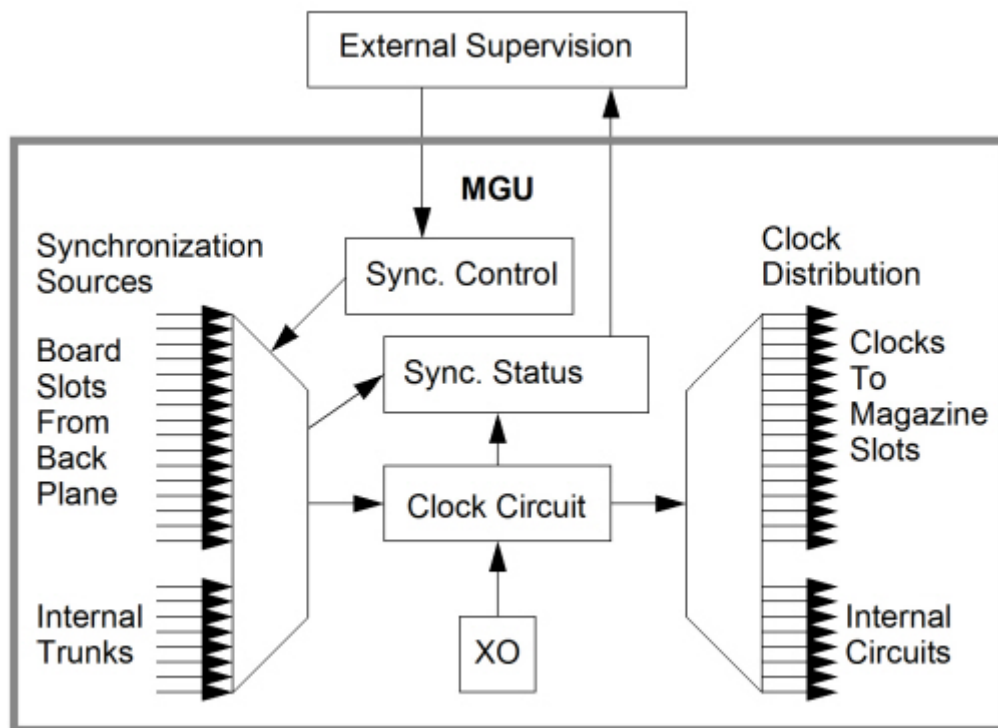
- The free running XO (Crystal Oscillator).
- One of the internal TDM trunks.
- Any board slot in a magazine (when a magazine is used) containing a TDM trunk board or other board that can provide PCM sync.

The clock circuit will automatically change to Holdover mode when the input signal is invalid, that is, the input is off by more than capture range which is more than ± 280 ppm or when the input signal is "gone" (steady high or low).

The locking time for the clock circuit is typically 50 seconds.

The on board FPGA routes the synchronisation clock from one of the trunks to the clock circuit unless the XO is used. The on board components and resource boards are then fed by the clock circuit and distributed by the FPGA.

Figure 2.5: Clock synchronization sources and distribution



Functionality

Restart

The MGU can be restarted by different sources:

- HW Reset Button (hole in front)
- HW Watchdog
- MX-ONE Service Node

The MGU has an external button (hole in the front) to make a HW reset of the board. When pushed the reset line to the Device Processor (DP) is pulled which causes an immediate restart of the DP and reboot of SW.

NOTE: Using HW reset should only be used as a last resort, since there is a risk of corrupting flash file system and/or configuration data.

In order to supervise SW execution there is a HW watchdog (in the FPGA), that is a timer that needs to be triggered by SW at regular intervals. If not triggered the DP reset will eventually be pulled.

The restart is logged (for fault analysis) in the linux syslog if ordered by MX-ONE Service Node or triggered by Watchdog.

Device Board Interface

The Device Board Interface provides a signaling and restart interface between MX-ONE Service Node and the Device boards in the 3U/7U subracks, and a PCM interface between Device boards and the TDM switch on MGU. This interface is also used for on-board provided functions that are implemented as “virtual boards”, like the Digital E1/T1 Trunk (see next section). Virtual boards is a way to emulate older Device boards but with new Hardware.

The MGU supports a propriety device board interface using 2Mbit HDLC and/or 128Kbit UART for signaling, and 2Mbit PCM for circuit switched media (32 x 64kbps timeslots) per device board position. Up to 16 device boards can be supported. The older UART protocol with slow signaling supported by DSU is not supported by MGU.

Digital Trunk E1/T1 Interface

MGU provides 8 digital trunk interfaces (PRI:s) of type E1 or T1. Each PRI is implemented as a “virtual board” in the “virtual magazine 2” (i.e. EQU range 2-0-00 to 2-70-31) and can be configured and run independently of each other.

The configuration of E1 resp T1 framing of each interface is done during “board” activation. Each interface can be configured as either:

- E1 with ISDN protocol (corresponding to a TLU76/11 board)
- E1 with CAS protocol (corresponds to a TLU76/13 board)
- T1 with ISDN protocol (corresponding to a TLU77/11 board)

MGU ISDN/PRI interface has been verified to comply with ETSI TBR004 (EU), TIA-968-A 47 CFR Part 68 (US), CS-03 Issue 9 Part VI (CA), AS/ACIF S038 (AU) and Newsletter No 125 (NZ).

MGU's Layer 1 and 2 supports ETS 300 011 and ETS 300 125 respectively.

When an E1 interface is configured as CAS trunk the ISDN Layer 2 protocol in time slot 16 is replaced by a CAS multi frame structure and signaling according to ITU-T G.732 for 2048 kbit/s. MGU supports MFC R1 and R2 signaling, replacing similar functionality in the MFU board.

Configuration parameters per PRI are set in run-time in the MX-ONE Service Node SW:

- Application = E1 or T1
- Network Termination = User side or Network side (for ISDN)
- A set of parameters unique for CAS trunk

NOTE: The E1 interface can as well be used for CAS Extension. In this configuration, 30s logical extensions can be represented at each E1 port.

Limitations

E1 CAS

MGU does not support all features and tones supported by the MFU board. MFU board(s) are therefore required where non standard MFC R1/R2 is used.

T1

Some of the counters and timers for error statistics reporting provided by TLU77/1 are not supported by the T1-PRI:s at MGU. Also the Facility Data Link (FDL) provided by TLU77/1 is not supported by the T1 PRI at MGU.

Layer 1 - Physical interface

The Layer 1 physical interface supports ISO 10173, and RJ45 connectors with wiring according to USOC RJ-48C.

The electrical connection is twisted pair 120 ohm for E1 and 100 ohm for T1.

MGU supports both European E1 and the North-American T1 TDM interface standard.

For E1 MGU provides automatic adaptation to "CRC-4 multi-frame structure" or "Double frame structure" for E1 and is specified in ETS 300 167, (based on ITU-T recommendations G704/G706).

For T1 or DS-1 MGU supports Super Frame (SF) or Extended Super Frame (ESF) framing scheme, bipolar with eight-zero substitution (B8ZS) or zero code suppression (ZCS).

Facility Data Link (FDL) is not supported and it is assumed that external equipment can be used, providing the FDL functionality.

Transmit Slip Buffer

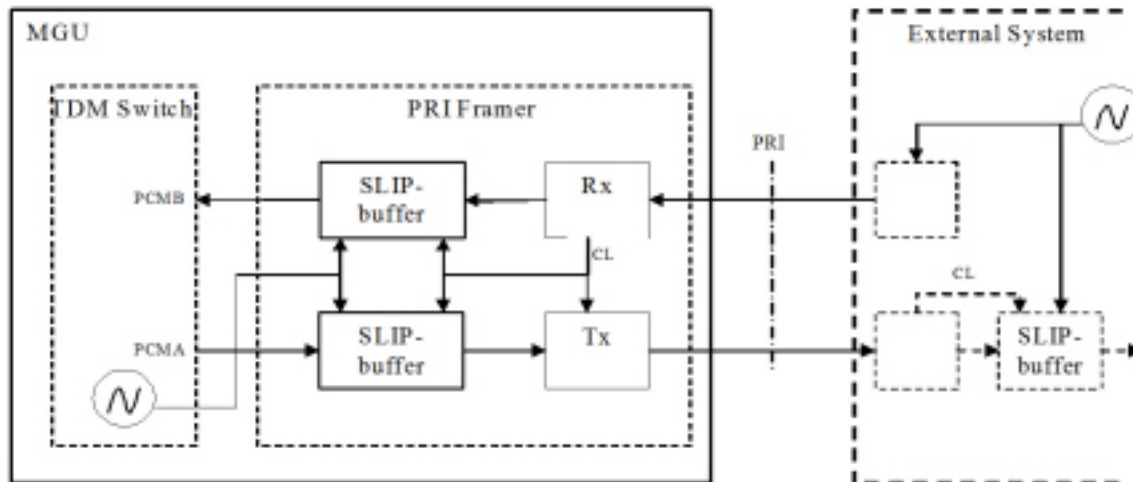
Each PRI interface contains "elastic" buffers (aka Slip buffers) in order to handle situations with different clock frequencies between MGU and an external telephony system. Slip buffers are always active in the receive path, but in the transmit path (towards the external system) it is optional to use.

In most scenarios when clock synchronization is properly administrated slip does not occur. However in some (temporary) scenarios, for instance when the DECT sync ring is used for PCM clock distribution

there are certain situations when it is preferred to use a free running MGU oscillator as PCM sync "master". In such situations to avoid SLIP alarms on "external systems" an activation of a Slip buffer in the transmit path of the E1/T1 Framer of MGU is needed.

The activation is made by setting the TX_Slipbuffer parameter (see ISDN parameters for details).

Figure 3.1: Block diagram showing the Slip buffers at the PRI interface



Layer 2 - Data Link Protocol

The PRI supports data link layer protocol ETS 300 125 (ITU-T ISDN user-network interface - Data link layer specification Q.921/I.441).

The PRI support point-to-point data link with non-automatic TEI assignment procedure.

Performance

The PRI signal capacity with an ISDN call rate of 5 calls/seconds generates less than 20% CPU load in net contribution from the ISDN traffic alone. The capacity figure is based on a ten Q.931 Layer 3 messages per call and one Layer 2 RR frame per message.

Voice Over IP

The MGU provides Voice over IP according to the RTP and SRTP (secure VoIP) protocols. MGU also supports DTMF relay in VoIP channels according to RFC 2833/RFC 4733.

VoIP channels are obviously used to convert media between SIP/H.323 terminals and circuit switched devices, but also to interconnect media gateways, known as "inter-GW calls". Thus, a "circuit switched" call between e.g. two analog phones in two different gateways may take a path over some VoIP channels, causing an additional round-trip delay of about 100 ms that might need to be considered.

The VoIP channels in MGU are dynamic resources in MSP and the amount of available resources depends on the actual MSP load and the configuration of particular channel. The MSP load for a particular VoIP channel varies very much depending on choice of codec, packetization interval, VAD/CNG, echo

canceler settings, crypto-suite, etc. For instance, use of VAD/CNG significantly reduce DSP load (and network bandwidth) and use of crypto-suite significantly increases load.

Furthermore, the selection of codec, packetization interval and crypto-suite also have great impact on the speech latency (especially packetization size). In general, latency sensitive installations should consider smaller packetization interval. Note that latency affects audio quality perception as well, especially when there is an echo situation

NOTE: VoIP channels is also used for inter-GW media (links between Media Gateways) and that configuration means of these is not same as configuring e.g. SIP/H.323 endpoints.

RTP

The Media Stream Processor encodes the PCM audio data from TDM timeslots (from TDM switch) into packets to the streaming interfaces, and decodes packets from the streaming interfaces to output PCM TDM line.

The audio coding (codec) standard is used for both the encoder and the decoder.

MGU supports Voice Activity Detection (VAD) and Comfort Noise Generation for all supported codecs. The VAD function can be fine tuned for emphasis on bandwidth saving or audio quality. Too high bandwidth saving might cause audible audio artifacts as choppy speech.

The following codecs are supported by MGU

- G.711 A- and μ -law, Appendix I (PLC) and II (VAD/CNG).
- G.729a with G.729 annex B (VAD/CNG).
- Clear Channel (bitwise exact transfer between TDM and IP, without echo canceler). Clear Channel is only intended for data or fax traffic. It shall never be used for voice, where echo canceling might be needed. Clear Channel uses dynamic payload type (PT in RTP header) as set by the MX-ONE Service Node from end-point negotiation

Selection of codec, PLC, VAD/CNG can be made on a per call basis from the MX-ONE Service Node based on SIP/H.323 media negotiation. MGU does not participate in any negotiation of media more than it reports its capabilities to the MX-ONE Service Node. It can be noted that use of VAD/CNG ("Silence Suppression") may lower DSP load and thus increase channel density (and also lower network load). On the other hand VAD/CNG might affect voice quality, so the use is a trade-off between density and voice quality. Unless the higher density or bandwidth saving is required it is a recommendation to disable VAD/CNG for best voice quality.

The MGU also supports Fax Pass Through. At modem and fax tone detection on the TDM side the MGU automatically switch to Pass Through mode using a predefined configuration which will set the RTP channel to G.711 codec and fixed jitter buffer, to be able to relay modem and fax data.

MGU allocates port numbers dynamically for RTP and RTCP from a port range that can be set by MX-ONE Service Node. RTCP port number is always RTP port + 1. When a new port number pair is allocated, always a pair with subsequent higher numbers is used. When highest configured port number is reached, the lowest is re-used again.

DTMF Detection and Relay in RTP Changes

Each RTP channel provides a DTMF detection and relay feature. In-band DTMF tones may be detected at incoming TDM side and may be relayed to packet side in one of three ways:

- Transparent, DTMF tones are passed as tone in the codec. This option is only useful when codec is G.711.
- As named telephone events (NTE) according to RFC 2833 / RFC 4733. In this mode in-band DTMF tones are removed from the TDM side and converted to events at the IP side (see also note below).
- Not relayed at all. Detected DTMF tones will be removed from the TDM side (see also note below).

The selection of detection and relay mode is made by MX-ONE Service Node based on e.g. SIP/H.323 negotiation with remote end-point/gateway or use-case. Note that DTMF detection can be enabled or disabled by MX-ONE Service Node, regardless of relay mode selected.

NOTE: When the RTP channel removes a DTMF tone there might be a leakage of less than 20 ms, i.e. may be audible but not detectable by any standard compliant DTMF receiver. By configuration it is possible to enable “complete removal” at the expense of longer channel latency (see chapter 4, *Configuration*). Complete removal is however only able to completely remove qualified DTMF digits.

Secure RTP (SRTP)

See the [Security](#) section.

Jitter Buffer

RTP packets sent over the IP network are subject to random variation in delays, out-of-sequence arrival, and a risk to be dropped. These artifacts decrease audio quality, and the jitter buffer is used to mitigate this. However, while the jitter buffer can improve audio quality it does this to the cost of increased voice delays. Long delays, especially in combination with echoes at far end (for example, caused by 2 to 4 wire hybrids in analogue lines) makes echoes more noticeable and disturbing. Although it might be tried to minimize delays in echo situations (if not the source of the echo could be removed) it must be understood that affecting voice quality due to dropped packets, which have negative impact on the echo canceler. See [Echo Canceled \(EC\)](#).

In MGU, the jitter buffer can be configured in adaptive or non-adaptive mode, and there are configuration parameters to adjust for actual network conditions.

NOTE: Configuration is per MGU and will affect all VoIP calls in that MGU, including inter GW media over IP.

The configuration of the jitter buffer will be a trade off between audio quality and delays. By default, the jitter buffer in MGU is adaptive with settings for a fairly “normal” network, to preserve audio quality over minimizing delays. For a very delay sensitive installation, where audio quality could be negotiated and/or network is very good, re-configuration might be considered.

Although primarily the jitter buffer is for adapting to artifacts caused by network, also VoIP endpoints (phones, gateways, proxies, and so on) is part of the network and can cause these. For example, soft SIP clients with no dedicated HW (e.g. DSP) for VoIP media will have substantially more jitter in outgoing RTP packets than a HW ditto. This can cause the jitter buffer to increase and thus to increase the delays even further. In those, and similar scenarios it might appear that the delay through MGU is longer than expected.

For non-voice calls over RTP, that is, Fax/Modem, it is recommended to have non-adaptive jitter buffer, which is automatically set if Clear Channel codec has been selected or if fax/modem tones have been detected in a voice call.

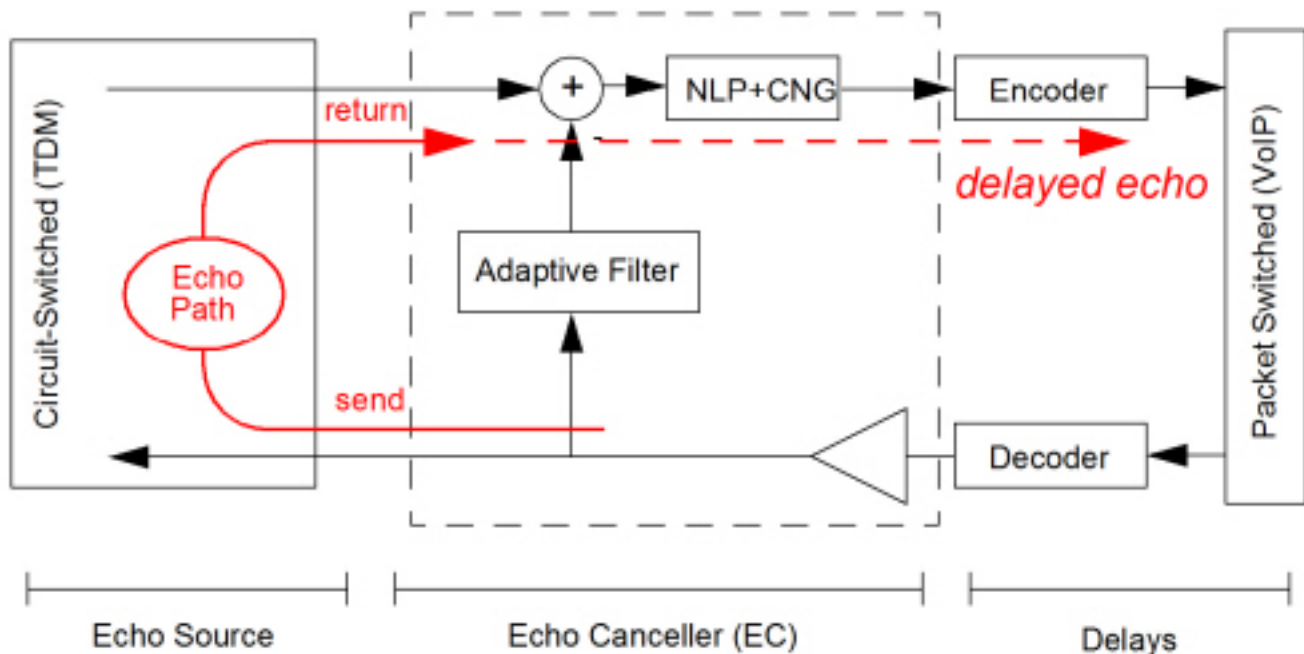
Echo Canceled (EC)

The Echo Canceled (EC) eliminates the possible echo of send signal from the return signal (see figure below). Echo is normally caused by reflections in transition from 2-to-4 wires, but also acoustic echo in telephones can occur. The EC is only used for calls over packet switched network (VoIP) as depicted in figure below.

There are various configurations of the EC which is described further on as well as in the See *Configuration*.

NOTE: Configuration is per MGU and will affect all VoIP calls in that MGU, including inter-GW media over IP. Thus, the choice of EC settings might have to be a compromise.

Figure 3.2: Schematic figure of Echo Canceled



The reason for using EC on VoIP calls is that echo in combination with (long) delays as caused by packet switched network is much more disturbing than echo when there is no or very little delay as can be expected in the circuit switched network (TDM).

The experience of how echo is perceived by the user on the packet side is a sum of echo source, echo canceler operation and packet network delays included delays in media gateways (encoding, decoding, jitter buffers), network (switches and routers) and endpoints (encoding, decoding, jitter buffers).

MGU supports two EC types for VoIP GW calls: Standard EC and Dual Filter EC (DFEC). EC type and settings very much impacts the MSP load.

NOTE: Changing EC type will cause a restart of the Media Control Application (MCA) and the MSP.

The EC also includes a Non-Linear Processor (NLP), a sub function that is capable of handling non-linear part of residual echo that the linear filter of the EC cannot cancel. By default, NLP is disabled and is only recommended to be enabled when needed.

When NLP is enabled and if engaged, it generates comfort noise (CNG) towards IP. CNG can also be changed to generate silence if CNG is not wanted.

Standard EC

The Standard EC echo tail length can be configured from 8 to 128 ms in 8ms steps, and the filter window is fixed to 24 ms. The advantage of the Sparse EC is MIPS savings which corresponds to higher channel density. The Standard EC can be improved by enabling Echo Path Change Detection (EPCD) that improves adaptation on filter window position.

NOTE: Setting EPCD will cause restart of Media Control Application.

Dual-Filter EC (DFEC)

The DFEC echo tail length can be configured from 8 to 128 ms in 8 ms steps. DFEC advantages:

- Avoids increased echo level caused by filter divergence during double-talk.
- Robust and fast detection of echo path changes.

The main disadvantage is that the DFE reduces DSP channel density, see section [Capacity and Limitations](#).

Standards Compliance

- The Standard EC without Echo Path Change Detection (EPCD) enabled is compliant with G.168/2002
- The Standard EC with EPCD enabled is G.168/2004 compliant for single reflection with echo dispersion less than 24ms and echo tail span less than 128 ms.
- The DFEC is G.168/2004 compliant for echo span up to 128ms (highest complexity).

A Few Recommendations in Case of Echo Problems

If disturbing echo is heard, the following actions can be recommended. (For information on parameters and settings, refer to See [Configuration](#)):

1. Analyze the traffic scenario to make sure echo canceler is turned on. Of special interest is the media path where the transition IP to TDM takes place (for instance, the Media Gateway where the call is routed to/from public ISDN trunk). Check that ClearChannel (ClearMode) codec is not used in RTP which always have EC turned off.
2. Turn on the NLP (Non-Linear Processor, a sub-function of the EC). By default NLP is disabled. The NLP will be able to take care of non-linear residual echoes that the linear filter cannot.
3. Turn on the DFE (Dual-Filter EC). This is a more complex EC algorithm and significantly improves echo canceling. Note however that DFE has some impact on MGU capacity, as further described in section [Capacity and Limitations](#).
4. If echo canceling is improved after re-configuration, but comes back again, make sure the configuration that was made is persistent, that is, parameters are marked "reload" in the MX-ONE Service Node and that proper data backup has been made.
5. Sometimes the echo delay path is longer than the echo canceler can handle (default is 64 ms, but can be changed up to 128 ms). The echo delay is the time from media (talker) is sent out on TDM and when it returns (attenuated) back to TDM. Usually, this is quite short, e.g. less than 10 ms but might be much longer in some situations. If the echo path is bigger than 64 ms but less than 128 ms the EC

filter lengths may be adjusted. Note however that a longer filter length will have some impact on MGU capacity.

SSRC Generation, Detection, and Collision Handling

For the outgoing (audio) RTP stream in a VoIP call, MGU creates a random 32-bit SSRC (Synchronization Source) value. This value is used for all RTP packets in that stream throughout the stream is active. If a call is put on hold, or any setting of the actual RTP stream is changed (for example, DTMF relay mode is changed) by the MX-ONE Service Node, the current stream is closed and replaced by a new. Hence, a new SSRC value is created for the new stream.

On corresponding incoming RTP stream, the MGU validates all received RTP packets. Packets with any SSRC value will be accepted as long as two packets with consecutive numbers and same SSRC are received. This allow the sender to change the SSRC value for a RTP stream. However, if the SSRC value changes too often during a shorter time period (about 1 second), this is referred to as a “SSRC violation”, and will cause the used RTP port to be blocked for a while to avoid violating port to be re-used. This situation is usually caused by two or more interleaved RTP streams towards same RTP port. If this happens there is probably a RTP sender that has not properly closed its RTP stream. In situations where such violations are expected for a longer time, the “ssrc_violation_filter_time” can be increased (or disabled), see [System Configuration Files](#) section.

Fax Relay T.38

The MGU supports ITU-T Recommendation T.38: Procedures for real-time Group 3 facsimile communication over IP Networks, ITU –T, June, 1998.

When in fax mode, the MGU stops encoding and decoding samples from the line as voice. Instead it encodes and decodes fax events according to the selected fax coding scheme, and passes these event indications over the packet network.

T.38 channels are dynamic resources in MSP and the amount of available channels depends on actual load of MSP.

Keycode Receiver

Keycode Receiver (KR) provide in-band DTMF and MFC (R1 and R2) detection on TDM timeslots from the TDM switch. MGU supports connecting KR in serial or parallel to the PCM stream. Serial connection is usually only applicable to DTMF tones, when in-band detection is used, e.g. for mobile extension calls.

Connecting KR in serial makes it possible to remove (see note below) the DTMF tones from PCM stream, but adds an extra latency of about 65 ms per default, which can cause echo problems in certain conditions. It is possible to have alternative settings to lower latency to as low as 16 ms at the cost of DSP channel density. See section [Keycode Receivers & Senders](#).

NOTE: When KR removes a DTMF tone there might be a leakage of less than 20 ms, thus may be audible but not detectable by any standard compliant DTMF receiver. Currently, there is no way to enable “complete removal” for KR as it is for DTMF receiver in VoIP channels.

In parallel KR connection, virtually no additional latency is imposed by KR, but DTMF tones are passed through the PCM stream.

Selection of serial/parallel connection is controlled by MX-ONE Service Node when KR connection is ordered.

KR is a dynamic resource in the MSP and number of available resources depends on actual usage and load of the MSP.

The KR in MGU is normally used in Mobile Extension (ME) calls. Currently, serial connection is used to by default setup KR for ME calls, thus each ME user will have an extra delay of about 65 ms in their speaking path.

Keycode Sender

The Keycode Senders (KS) provides in-band DTMF and MFC (R1 and R2) sending on TDM timeslots to the TDM switch. The duration (1-255 ms, or continuous) and attenuation (0-36dBm0) of the DTMF and MFC key codes is determined by the MX-ONE Service Node in the start of the KS resource and attenuation setting in the TDM switch.

KS is a dynamic resource in the MSP and number of available resources depends on actual usage and load of the MSP.

Tone Senders

The Tone Senders (TS) provides in-band call progress tone generation on TDM timeslots to the TDM switch. Tone characteristics are defined in tone configuration files, one per supported market (see section [Market Files](#)). Note however that attenuation level of a tone might be affected by the attenuation in the TDM switch connection established by MX-ONE Service Node.

TS is a dynamic resource in the MSP and number of available resources depends on actual usage and load of MSP.

There are however a few Tone senders that are permanently setup by the MX-ONE Service Node at start-up. These can be connected later at demand through sun-fan connections to several receivers.

Recorded Voice Announcement

MGU provides locally stored Recorded Voice Announcements (RVA) to be played out on demand by a Media player channel in MSP on timeslots connected to TDM switch. The local storage area (i.e. the flash disk) on MGU allows about 60 minutes of media files, corresponding to about 30Mbyte, to be stored.

The supported media file format is WAVE audio, ITU-T G.711 A-law/ μ -law, mono 8000 Hz. Normally, A-law is used on a A-law market, and μ -law is used on a μ -law market, but MGU allows the media files to be stored in any of these, independent of market.

The RVA Management function is located in Operation & Maintenance Application (OMA) subsystem. The function is responsible for the RVA Management interface to Service Node (SN) and to download files from a Web server. The RVA Management informs (Media Control Application) MCA when new RVA files have been downloaded and activated. The maximum file size (current default value is 30Mbyte) and maximum number of files (current default value is 250) that can be downloaded to the flash disk can be

changed. After downloading to flash disk, MCA take care of downloading (activating) files into MSP external SRAM. The activation order is received from OMA after download to file system.

NOTE: Download and activation of RVA files is done by very low prioritized tasks in order to minimize impact on run-time traffic, thus might take various time to finalize depending on actual traffic load conditions.

Run-time handling of RVA messages (Media player sessions) are then handled by MCA on demand from MX-ONE Service Node.

The Media player channels are dynamically allocated from MSP by MX-ONE Service Node orders, and the maximum achievable density depends on the actual MSP load. One Media player load is roughly about twice the load of a G.711/20ms RTP channel.

Media files can be played out on one TDM timeslot or many using the TDM switch multi-cast function (sunfan). A media file can be played once or in repeat, as controlled by MX-ONE Service Node orders. At media file upgrade all running media player sessions will be terminated, informing MX-ONE Service Node by a play ended event. Sessions that are playing in repeat, need to be re-started by MX-ONE Service Node.

Media Streaming with MGU

MGU does not provide media streaming, but can receive plain RTP stream from Media Servers to provide media streaming for legacy device boards.

Media streaming is used for MOH, MOW and RVA, thus media files is not required to install on MGUs when media streaming is enabled. Refer *MEDIA STREAMING* section in MX-ONE Media Server description for more information.

IP Network Redundancy and Security

General

The MGU is supporting IPv4 and it is also supporting IPv6 in the MX-ONE 6.0 release or later versions. The IP version configuration combination can be IPv4 only or both versions IPv4 & IPv6. The factory default configurations do not contain any IPv6 settings except for the IPv6 Local Link (LLC) address which will be set at boot up.

NOTE: This means that if you want to use security (encryption) you cannot use IPv6.

The control interface and the media interface parameter for IPv6 should be configured using MX-ONE Service Node's media gateway control and media gateway interface commands.

The NV parameter names for these interfaces are:

- eth0_ip6 (control interface)
- eth2_ip6 (media interface)

NOTE: The LLC address is only visible in the linux network configuration information and not visible in the MGU's NV parameter area.

The MGU also support 2 types of network redundancy:

- Switched (Ethernet) redundancy (Link Fail Over).

- Subnet (IP) redundancy

NOTE: Subnet redundancy is not supported in MX-ONE release 6.0 or later version.

The MGU also supports Server redundancy by allowing a standby server to take over the MGU from the ordinary server.

The MGU supports both media security (SRTP) and IP signaling security (IPsec).

GARP and Unsolicited Neighbor Advertisement for Media

Starting from MGU version 2.9.0.2 and later versions, the MGU supports Gratuitous ARP and Unsolicited Neighbor Advertisement (NA) for ongoing VoIP and FoIP calls. When it is received, the MGU will redirect the stream to the new destination MAC-address for the affected VoIP and/or FoIP calls.

Switched Redundancy (Link Fail Over)

Link Fail Over (LFO) is handled by the MGU's internal switch. The Media Stream Processor's (MSP) and the Device Processor's (DP) Ethernet traffic will be switched towards one of the LAN ports (active LAN ports). At fail over the traffic will be redirected towards the redundant LAN port and the MGU will stay in that state until a new failure appears. However, there is a period of about 30 seconds after link switch before link monitoring starts again.

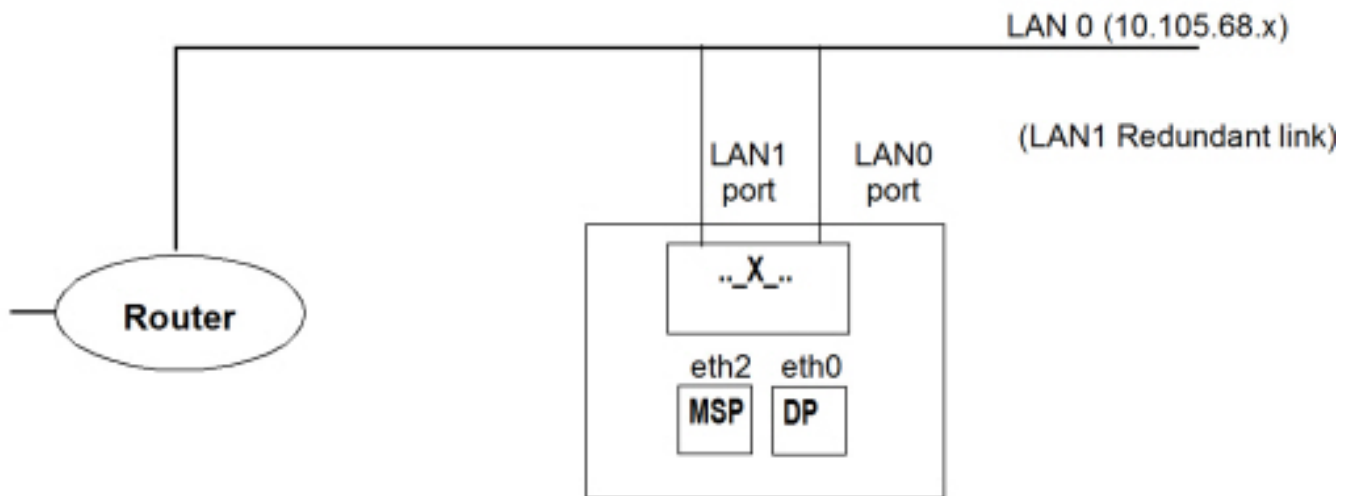
LFO is activated when both LAN ports are connected.

Link switch will occur when a link suddenly disappears or if no control signalling messages are received and the default gateway is not reachable. An alarm will be raised at this condition. The alarm will be cleared when the previously active LAN port has link again.

MGU can also be configured to raise an alarm if the link on the passive LAN port is lost while the active LAN port is operational. In this case the alarm is cleared when the link on the passive LAN port has link again. If alarm is indicated for passive or active LAN, then there might be a network related issue for further investigation. See further *Installation Instructions - Installing and Configuring*, section [MGU Board Setup](#).

NOTE: At link switch, LAN1 port will inherit the PHY configuration from LAN0 port. It is not supported to have different PHY configuration on the LAN ports when LFO is used.

Figure 3.3: Switched redundancy

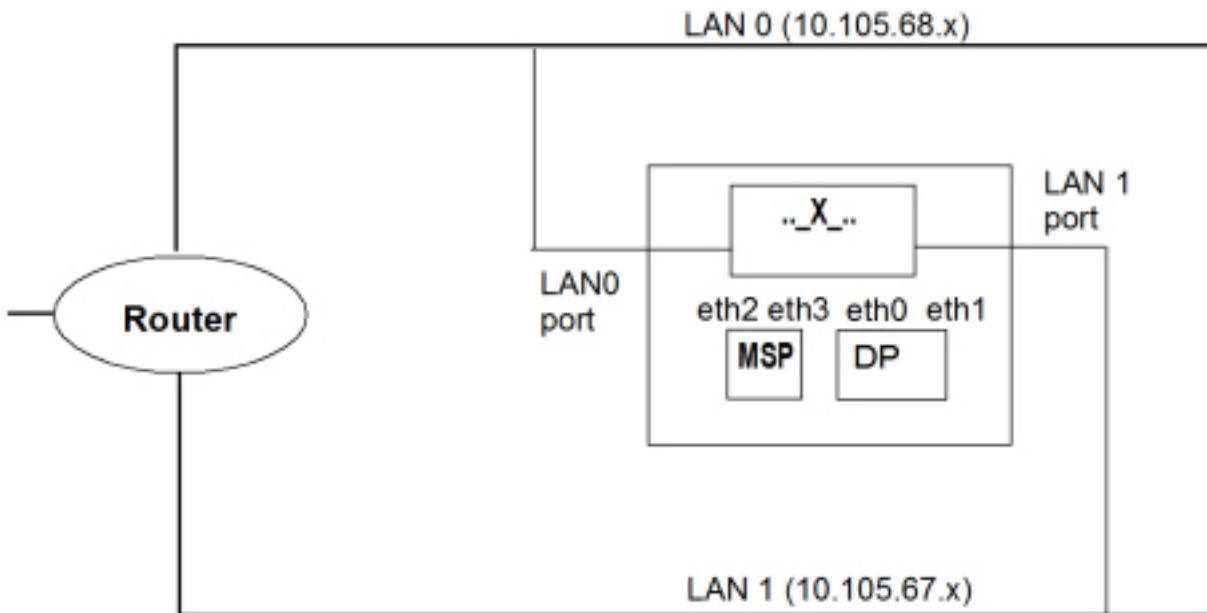


Example:

eth0_ip = 10.105.68.60
eth2_ip = 10.105.68.61

Subnet Redundancy (Network Redundancy)

Network redundancy is handled by a supervision process on the MGU. When the default gateway is not reachable on the present LAN the MGU will switch subnet and reconfigure the Media Stream Processor (MSP) with a new IP-address (ex: eth3-IP) specified by the configured MGU network parameters and the switch will stream the media packet towards redundant LAN (ex LAN1).

Figure 3.4: Subnet Redundancy

Example:

```
eth0_ip = 10.105.68.60
eth2_ip = 10.105.68.61
eth1_ip = 10.105.67.60
eth3_ip = 10.105.67.61
```

Server Redundancy

There is no specific built in support for server redundancy other than the MGU allows another (standby) server to take over the MGU when the connection to currently connected (ordinary) server is lost.

Only one server can control the MGU at the time, so the standby server will take over all MGU native resources as well as device boards in the MGU magazine. However, there is no support to synchronize between server and MGU, thus when a standby server takes over, all MGU activities will have to be restarted by the server to take MGU and server to a common state (i.e. closing RTP sessions, reset TDM switch connections, and restart virtual ISDN boards and device boards).

During server downtime all media established before server fall out is kept to allow for call continuity during downtime, although, this will not guarantee that calls are not disconnected by a remote side. Note also that ALL calls will be disconnected when standby server comes up.

NOTE: MGU actually cannot distinguish between server failure and normal disconnection. Thus, MGU behavior is the same regardless of how disconnection from MGU is made.

Port Authentication using 802.1X

IEEE 802.1X is an IEEE Standard for Port-based Network Access Control (PNAC).

It is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN or WLAN.

802.1X authentication involves three parties: a supplicant, an authenticator, and an authentication server.

The supplicant, a client that provides credentials to the authenticator, is a client device that wishes to attach to the LAN/WAN. The authenticator is a network device, such as an Ethernet switch or wireless access point; and the authentication server is typically a host running software supporting the RADIUS and EAP (Extensible Authentication Protocol) protocols. 802.1X uses EAP for message exchange during the authentication process. With EAP, an arbitrary authentication method, such as certificates, smart cards, or credentials, is used.

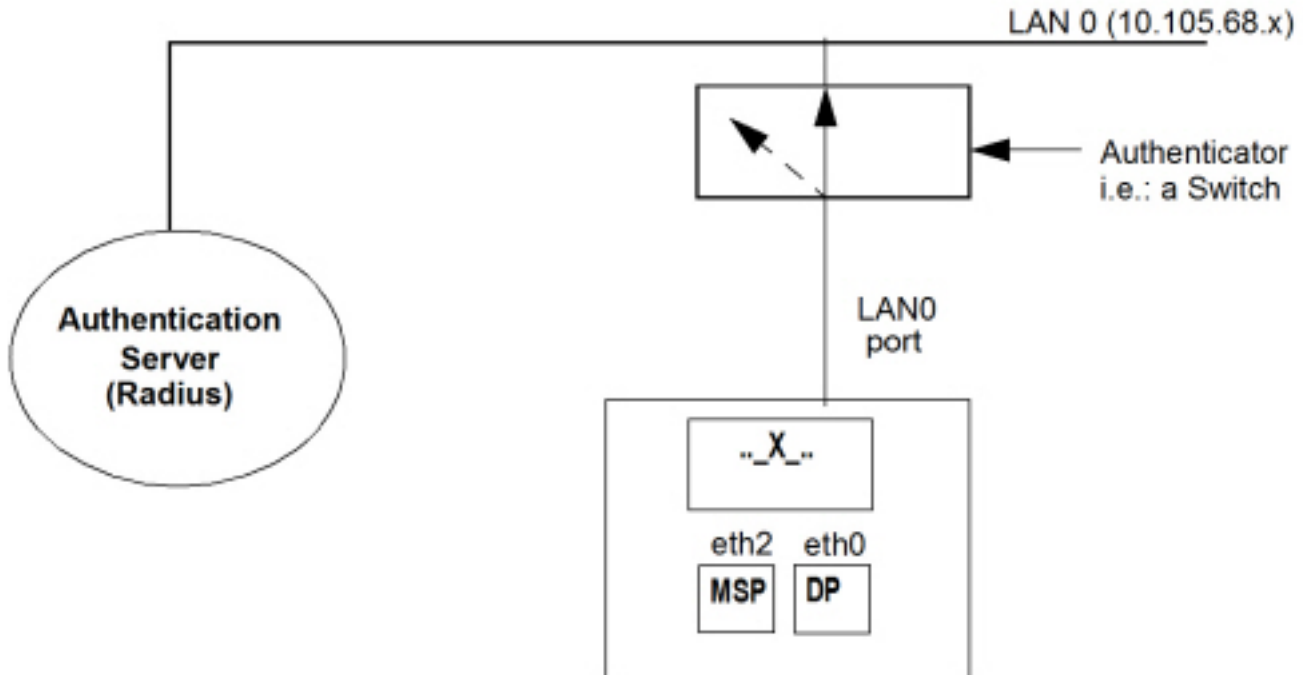
The most common EAP methods are EAP-TLS, EAP-TTLS and EAP-PEAP authentication.

- **EAP-TLS** is an IETF open standard, and is well-supported among wireless vendors. It offers a good deal of security. It uses PKI to secure communication to the RADIUS authentication server which provides very good security.
- **EAP-TTLS** is widely supported across platforms, and offers good security, using PKI certificates only on the authentication server, with tunneled EAP or PAP/CHAP/ MSCHAP/ MSCHAPV2 authentication.
- **EAP-PEAP** is similar in design to EAP-TTLS, requiring only a server-side PKI certificate to create a secure TLS tunnel to protect user authentication, with tunneled EAP authentication.

With 802.1X port-based authentication, the supplicant provides credentials, such as user name/password or digital certificate, to the authenticator, and the authenticator forwards the credentials to the authentication server for verification. If the authentication server determines the credentials are valid, the supplicant (client device) is allowed to access resources located on the protected side of the network.

The Media Gateway supports 802.1x over wired LAN with EAP-TLS as the supported authentication method. The switch port to which the Media Gateway Unit is connected must be configured for 802.1X authentication for multiple hosts. That is, you must be able to connect multiple hosts to this single port for 802.1X authentication. When one client (MGU eth0 - signaling) is authenticated, all the other clients (MGU eth1 -media) are also granted access to the LAN.

The picture below shows port access when the port is unauthorized (dashed line) and when the port is authorized.

Figure 3.5: Port Authentication 802.1X

Security

Media Security - Secure RTP (SRTP)

MGU provides VoIP security according to the SRTP protocol (RFC 3711 and RFC 6188), using data flow encryption with AES in Counter Mode (CM) and authentication with HMAC-SHA1.

Data Flow Encryption

For encryption and decryption of the data flow, SRTP standardizes utilization of only a single cipher, Advanced Encryption Standard (AES), which can be used in two cipher mode: Integer Counter Mode (CM) or F8 Mode. Only CM is supported in MGU.

Authentication

AES algorithm does not secure message integrity itself, to authenticate the message and protect its integrity, the keyed-Hash Message Authentication with Secure Hash Standard (HMAC-SHA1) algorithm is used.

Key Derivation

In SRTP, the different keys used in a crypto context (SRTP encryption and salt keys, and SRTP authentication key) is derived from one single Master Key (per media direction). That is, from the master keys all the necessary session keys are generated by applying the key derivation function. MGU derives the Master key for the transmitted SRTP stream from a high entropy random source. The Master key for received SRTP stream (derived by remote end-point or gateway) is received from the MX-ONE Service Node. The master keys are derived only once before the call set up. Re-keying is thus not supported.

IP Signaling Security (IPsec)

MGU supports Internet Protocol Security (IPsec) that can be used to secure IP communication between MGU and a remote IPsec peer (a remote Service Node or Gateway/ Firewall). For example, IPsec may be used in a branch node scenario where Service Nodes are located in a head office and MGUs in branch offices while signaling over the Internet. IPsec is supported for MGU signaling over IPv4 networks.

Figure 3.6: IP communication between MGU and a remote IPsec peer



The IPsec protocol suite is an open standard described in RFC 2401. IPsec is based on the following protocols:

- Authentication Header (AH) to provide connectionless integrity, data origin authentication, and an optional anti-replay service.
- Encapsulating Security Payload (ESP) to provide confidentiality (encryption), and limited traffic flow confidentiality. It also may provide connectionless integrity, data origin authentication, and an anti-replay service.
- Internet Key Exchange protocol (IKE) to provide automatic establishing of Security Associations (SA) and peer authentication. MGU supports IKE version 1 (IKEv1). IKE is only used in automatic key management procedures.

AH and ESP may be used in combination, but usually only one of them is used. When only ESP is used it is usually in combination with authentication. By default MGU enables ESP with authentication. Most importantly, ESP will protect the confidentiality of the SRTP keys exchanged between Service Node and MGU for secure RTP. For some installations it might make sense to only use AH.

IPsec may be used in tunnel (a.k.a VPN tunnel) or transport mode. In tunnel mode, the entire IP packet is protected by IPsec. This means IPsec wraps the original packet, encrypts it, adds a new IP header and sends it to the IPsec peer. In transport mode, only the IP payload is protected by IPsec. Tunnel mode is usually used between networks or host-to-networks, and transport mode is usually used end-to-end between two hosts. MGU creates a transport if a destination address is specified or a tunnel if destination network is specified.

MGU allows automatic IPsec SA establishment and peer authentication through IKE/ISAKMP using either pre-shared keys or RSA signed digital certificates. It is also possible to use manually configured SAs, but is usually not recommended.

It is possible to setup multiple IPsec tunnels/transport between MGU and remote peers.

For more information about how to setup IPsec, see Operational Directions *MGU SECURITY CONFIGURATION*.

Visual Indications

There are a few LEDs on the MGU front showing operational status of the board and interfaces. There is the main Status LED and additional LEDs on each Ethernet and E1/T1 port.

Status LED

The Status LED on MGU shows general operating status as follows:

- **Flashing red** - The board is in boot loader mode.
- **Steady red** - The board is not active.
- **Steady green** - The board is activated (connection with MX-ONE Service Node established on all three communication ports).
- **Flashing green** - The board is activated, and signaling packets (SCTP) are transmitted between MX-ONE Service Node and MGU.

Ethernet LEDs

- **Green LED** - Shows flashing green when ethernet packets are sent or received
- **Yellow LED** - Shows steady yellow when 100Mbit

E1/T1 LEDS

- **Green LED** - Shows steady green when layer 1 is activated and there is incoming frame synch. Flashes when there is layer 2 activity, i.e. HDLC data frames are sent or received.
- **Yellow LED** - Shows steady yellow if the link is PCM synchronization source.

Alarm Handling

Each alarm type has a unique number and an alarm text which is sent in a status message to MX-ONE Service Node. Alarms are also logged in the MGU's syslog at `/var/log/syslog` in MGU's file system.

The alarm table below enumerates all alarms that can be generated from the MGU board, with their alarm number and severity. These alarms belongs to the Media GW alarm domain (5) in MX-ONE Service Node.

Table 3.1: Alarms. The Media Gateway alarms and Alarm encodings. All Media Gateway alarms belongs to "alarm domain" 5 in Telephony System.

Encoding *)		Alarm Description	Severity
Code	ID	-	-
9	9	Power Problem: -5V failure in backplane	Critical
9	10	Power Problem: +5V failure in backplane	Critical
9	11	Power Problem: -12V failure in backplane	Critical

Table 3.1: Alarms. The Media Gateway alarms and Alarm encodings. All Media Gateway alarms belongs to “alarm domain” 5 in Telephony System.

9	12	Power Problem: +12V failure in backplane	Critical
18	18	Equipment Malfunction: External alarm A raised **)	Alert
19	19	Equipment Malfunction: External alarm B raised ***)	Undetermined
20	20	LAN Error Lost connection to LAN 0	Warning
20	21	LAN Error Lost connection to LAN 1	Warning
22	22	LAN Error: LAN0 gateway unreachable	Warning
22	23	LAN Error: LAN1 Gateway unreachable	Warning
24	24	VLAN Error: VLANs with same GWMAC ADDR	Warning
34	26	MGU Resource Warning: High load in DSP core inside MSP	Warning
34	27	MGU Resource Warning: High load in ARM core inside MSP	Warning
34	28	MGU Resource Warning: No more time slots available	Minor
34	29	MGU Resource Warning: MSP out of resources	Minor

*) In Telephony System’s alarm log, only the alarm code (and domain) is logged (e.g. 5 and 22 for LAN Error). In MGU’s syslog, only the alarm Id is logged, as for example: “AlarmSupervisor-raiseAlarm for id = 23 MO = MGW” indicating LAN1 Gateway unreachable.

**) The alarm definition (alarm code and alarm text) is configurable from MX-ONE Service Node to describe actual alarm source.

***) The alarm definition (alarm code and alarm text) is configurable from MX-ONE Service Node to describe actual alarm source.

Configuration

Most configuration of MGU is made through MX-ONE Service Node and/or MX-ONE Managers and sent to MGU via its communication interface.

For normal operation there is no need to change any of the settings described here. Changing these settings should only be made if advised by Mitel Service Technician.

Exception to this is the IP address for LAN0 which is required to setup at installation time.

NOTE: MGU generally don't validate configuration parameters, thus it is important to check that reasonable values are used

Restriction of SSH Access to MGU

In previous version of the MGU, there was no restriction for users who can access the MGU through SSH login if they have login credentials.

From MGU version 2.9.0.1, where you can limit the access for remote SSH login users. By executing the `mgw-setup` command, you can configure the allowed host addresses and/or networks for both IPv4 and IPv6. This command can only be executed with root access.

Use the below command and follow the on-line instructions:

- `mgw-setup --set-allowed-hosts`

A valid host address (a single address) is available either in IPv4 or IPv6 format and must not be in the Classless Inter-domain Routing (CIDR) notation.

For Example:

- 192.168.100.121 - Valid.
- fec1::1010:1:1::121 - Valid.
- 192.168.100.200/24 - Not a valid single address.
- fec1::1010:1:1:121/64 - Not a valid single address.

A valid network address, that is, a range of host addresses must be in the Classless Inter-domain Routing (CIDR) notation, where the last bits of the host address must be zero.

For example:

- 192.168.100.0/24 - Valid (192.168.100.0 – 192.168.100.255).
- 192.168.100.128/24 - Not a valid network range since the last 8-bits not in the IP-address not is zero.
- fec1:1010:1:1::/64 - Valid (fec1:1010:1:1:0:0:0:0 – fec1:1010:1:1:ffff:ffff:ffff:ffff).
- fec1:1010:1:1::99/64 - Not a valid network range since the last 8-bits not in the IP-address not is zero.

NOTE: Entering of any wrong addresses can lock the MGU from all SSH access. The only way to gain access is through the console via USB or serial port.

Following two files containing the parameters are:

- `/etc/mgw/system/hosts_allowed_ipv4`
- `/etc/mgw/system/hosts_allowed_ipv6`

NOTE: Do not edit these above files, these are automatically generated.

To allow full access to the MGU, these files must be removed. This command is executed with root access.

- `rm /etc/mgw/system/hosts_allowed_ipv4`
- `rm /etc/mgw/system/hosts_allowed_ipv6`

Boot Parameters

The boot parameters are mainly to boot the operating system and are stored in non-volatile memory on the board. Although these parameters can be changed both in boot loader mode and linux mode (i.e. the mode that will be entered after board has restarted), it shall never require to modify them from boot loader.

In linux mode the boot parameters can be changed through the Maintenance port (USB) using a serial (V.24) dongle at 9600 baud, no parity, 1 stop bit. Only serial dongles using the PL2303 device are supported. It can be noted that the USB port is not active in boot loader mode.

It is also possible to change parameters through SSH login (but obviously, the IP address need to be known then).

NOTE: The only settings required by end-user is the IP address and default router for LAN0, i.e. parameters `def_route` and `eth0_ip` (IPv4), and `def_route6` and `eth0_ip6` (IPv6). Other parameters are set indirectly through MX-ONE Service Node commands. Faulty settings or removing parameters might cause malfunction of the MGU.

NOTE: There is no check that boot parameters are spelled correctly (a misspelled parameter will create a new parameter, not recognized by applications), thus check spelling if change doesn't take effect.

Market/Site Parameters (attributes)

The MGU's management application (OMA) maintains a set of configurable parameters in an internal (non-persistent) data base in the application. This data base is initially loaded with default and market specific values from MGU's file system (see also [Market Files](#)). The valid market is selected by MX-ONE Service Node command `mxone_maintenance`.

Each parameter can also be viewed and set with the `media_gateway_info` command from the MX-ONE Service Node to override the default or market settings.

Any time the data base is changed, OMA will push out changes to other MGU applications. In addition, if an application is restarted it will request settings from OMA. A restart of OMA (or MGU) will reload data base from files and cause MX-ONE Service Node to send all changed parameters again.

Below is a brief description of all parameters (grouped functional wise).

NOTE: Be aware of that some parameters will cause restart of MGU applications in order to take immediate effect. Thus, it is only recommended to change these during maintenance hours.

VoIP

Table 4.1: RTP parameters (RTP MO)

Name	Value Range	Description
ComfortNoiseGeneration		Parameter is not used
JB_adaptionPeriod	1000..65535 ms	Controls the speed at which the jitter buffer can adapt downwards when current network conditions allow. Default is 10 s, which can be set lower for good networks.
JB_delayInit	0..200 ms	Initial delay of jitter buffer. Default is 0 ms.
JB_delayMax	0..200 ms	Controls maximum size of jitter buffer. Default is 200 ms. If "Hardmode" is selected this is the maximum size jitter buffer can grow. If "Softmode" then deletion occurs at "JB_deletionThreshold".
JB_delayMin	0..200 ms	Controls minimum size of jitter buffer. Default is 0 ms.
JB_deletionMode	0..1 (boolean)	0=Softmode (audio quality focus, default) 1=Hardmode (delay focus)
JB.deletionThreshold	delayMax..500 ms	Packets exceeding deletionThreshold are deleted. Default is 500 ms.
PacketLossThreshold		Parameter is not used
VADTune	0..4	Controls VAD threshold to improve bandwidth (low value) or improve voice quality (high value). Too low value might give undesirable impact on voice quality. It is recommended to set at least 1 (default)
VLANTagValue	0..4095	VLAN ID for RTP packets (0 disables VLAN tagging)

NOTE: Setting delayMin = delayMax = delayInit makes jitter buffer non-adaptive

Table 4.2: VoIP channel DTMF detection parameters (TDM MO)

Name	Value Range	Description
DTMF_CompleteRemoval	0..1 (boolean)	0 = Detected DTMF tones (valid DTMF digits) are removed, but can be audible 1 = Detected DTMF tones are removed totally (adds channel delay)
DTMF_MinToneOnTime	0..8191 ms	Min tone length to qualify as DTMF digit
DTMF_MinToneOffTime	0..8191 ms	Separation between digits
DTMF_MaxDropoutTime	0..8191 ms	Max tone dropout to qualify as one digit

NOTE: There are further DTMF parameters that can be set with AUXKR MO (see table 9, note 2).

NOTE: To meet the most industry standards, default settings minToneOn=30ms, minToneOff=35 and minDropoutTime should be used.

Table 4.3: VoIP channel Echo Canceller parameters (TDM MO) (Sheet 1 of 2)

Name	Value Range	Description	EC Type
EC_ECType	0..1	Selects Echo Canceller type: 0 = Standard Echo Canceller (STD) 1 = Dual-Filter Echo Canceller (DFE)	STD/DFE
EC_DFECFilterSize	8..128 ms (in steps of 8 ms)	Filter length for DFEC	DFE
EC_DFECMinErl		DFEC Minimum ERL setting (do not change)	DFE
EC_DFECAttenuation		DFEC Rx output digital gain (do not change)	DFE
EC_ECCrossCorrelationCalculation		Not used	-
EC_EchoPathChange	0..1 (boolean)	0 = Disable EPCD 1 = Enable EPCD	STD
EC_ErlChangeDetection		Not used	-
EC_FastConvergenceControl	0..1 (boolean)	Accelerates filter convergence for long filters	STD
EC_ECWindowSize	24 ms	EC window size for Standard EC	STD

Table 4.3: VoIP channel Echo Canceller parameters (TDM MO) (Continued) (Sheet 2 of 2)

Name	Value Range	Description	EC Type
EC_NLPControl	0..1 (boolean)	0 = Disble NLP 1 = Enable NLP	STD/DFE
EC_NLPTune	0..2	Not used.	STD
EC_ECTailLength	8..128ms (in steps of 8 ms)	Filter length for STD EC	STD
EC_EchoCancellerEnable		No used	-
EC_CNGenable	0..1 (boolean)		STD/DFE
SilenceToPCMInterface	0..1 (boolean)	Not used (parameter is controlled indirectly by CNG settings)	-

NOTE: The column “EC type” tells for which EC type the parameter is valid.

NOTE: Changing EC parameters from default values might lower VoIP channel and other DSP resource densities

Digital Trunks

Table 4.4: ISDN parameters, common to all trunks (ISDN MO) (Activation of TX Slipbuffer can be done on each individual trunk) .

Name	Value range	Description
Freebits	0..127	Default = 127
CRC_Threshold		Not used
N2x4		Not used
K	0..127	
N200	1..10	Default = 3
N201	260	Do not change
T200	1000..2000 ms	Default 1000 ms
T203	5000..20000 ms	Default 10000 ms
TX_Slipbuffer	0-255	Activation of TX elastic buffer. Default 0 = Off for all PRI:s. The parameter is bit oriented, see the table 7 below.

Table 4.5: Tx_Slipbuffer

Binary value	Decimal value	PRI to enable Tx elastic buffer
00000001	1	PRI 0
00000010	2	PRI 1
00000100	4	PRI 2
00001000	8	PRI 3
00010000	16	PRI 4
00100000	32	PRI 5
01000000	64	PRI 6
10000000	128	PRI 7
01000010	66	PRI 1 & PRI 6
11111111	255	PRI 0 to 7 (all)

NOTE: There are also configuration parameters per PRI/trunk that are set in run-time from MX-ONE Service Node. Physical interface = E1 or T1. User Network = User or network side.

NOTE: Changing the value for the Tx_Slipbuffer parameter requires the corresponding “virtual board” (PRI interface) to be restarted in order to take effect.

NOTE: Activation of Tx_Slipbuffer on PRI shall not be used when the PRI provides synchronization to a remote system.

Tone Senders

Table 4.6: Tone Sender parameters (AUXTS MO)

Name	Value range	Description
N/A	N/A	N/A

Keycode Receivers & Senders

Table 4.7: Keycode Receiver parameters (AUXKR MO)

Name	Value range	Description
AntiTapDetection		Do not change
DetectionDelay (see note 2)	18/30/40 ms	Specifies the minimum on time of DTMF signals to be detected as valid digits.
DtmfRemovalLevel	0..2	Not Used
EarlyDetection		Not Used
FrequencyDeviation (see note 2)	15..25 (1.5-2.5%)	Frequency deviation of DTMF tone pair
MaxAntiTapToneDropoutTime		Do not change
MaxDropoutTime	0..8191 ms	Max tone dropout to qualify as one digit
MaxToneDropoutTime		Do not change
MinAntiTapToneOffTime		Do not change
MinAntiTapToneOnTime		Do not change
MinLevelThreshold (see note 2)	140-480 (-14 to -48dBmo)	Min threshold level for DTMF frequency components
MinToneOffTime	0..8191 ms	Min tone off length to qualify as digit
MinToneOnTime	0..8191 ms	Min tone length to qualify as DTMF digit
NegativeTwist (see note)	10..160 (1-16dB)	Negative Twist
PacketSize	5,10,20,30,40,50, 60 ms	Changes channel latency for DTMF receiver at the expense of channel density (30 ms is default).
PositiveTwist (see note 2)	10..160 (1-16dB)	Positive Twist
SnrThreshold (see note 2)	-30..60 (-3 to 6dB)	Signal to Noise ratio. 16 bit value. Negative values are specified as sign bit + value, e.g. 32768 + 30 = -3.0 dB.
ZeroInterDigitDetection		Do not change

NOTE:

1. To meet the most industry standards, default settings should be used.
2. These parameters are also applicable to DTMF receivers in VoIP channels.

Table 4.8: Tone Sender parameters (AUXTS MO)

Name	Value range	Description
N/A	N/A	N/A

Multi Party (Conference bridge)**Table 4.9:** Multi Party (Conference) parameters (AUXMP MO)

Name	Value range	Description
AGCEnablePCMtoMixer	0 = Disable AGC 1 = Enable AGC	Automatic Gain Control. Not Supported.
AGCPCMtoMixerMaxGain		Not Used.
AGCPCMtoMixerMinGain		Not Used.
AGCPCMtoMixerRate		Not Used
AGCPCMtoMixerTargetLevel		Not Used.
HighAttenuation	0 .. 14 dB	Attenuation applied to participants when number of participants in a conference is above "ParticipantThreshold" (default 6dB).
LowAttenuation	0 .. 14 dB	Attenuation applied to participants when number of participants in a conference is below or equal to "ParticipantThreshold" (default 3dB)
ParticipantThreshold	3 .. 15	Determines the number of participants in a conference when low or high attenuation shall be applied (default 4)

Recorded Voice Announcements

Table 4.10: Voice Sender parameters (AUXVS MO)

Name	Value range	Description
N/A	N/A	N/A

Quality of Service (QoS)

Table 4.11: Quality Of Service parameters (QOS MO)

Name	Value range	Description
TypeOfServiceForMedia	bit mapped (decimal value)	The ToS field in the IP header for RTP packets. Default value is 184 (decimal). Refer also to RFC 2474.
TypeOfServiceForControl	bit mapped (decimal value)	The ToS field in the IP header for control signaling between MX-ONE Service Node and MGU services. Default value is 152 (decimal). Refer also to RFC 2474. NOTE: Changing this parameter causes restart of MGU services.

Market Files

The /etc/mgw/markets directory on MGU contains files with default configuration settings and call progress tone characteristics for all supported markets. There is one common file with default settings for all markets and one file per unique market where differences compared to the default settings is stored. At MGU startup, the default file is read into an internal data base, and when MX-ONE Service Node orders MGU to select market, the corresponding market file and tone characteristics file is loaded, updating the data base.

NOTE: Some parameters in the market files might require or cause restart of MGU applications if different from default file, but normally market is not changed in run-time.

NOTE: Market settings changed by the MX-ONE Service Node command `media_gate-way_info` only updates the internal data base rather than writing changes to the market files.

System Configuration Files

In the /etc/mgw/system directory are stored some MGU system configuration files, *.conf. These contains settings for the MGU SW applications, which are read and used when the applications starts or re-starts. Thus, if any setting is changed, then the applications need to be restarted for the new setting to take effect.

NOTE: Some settings are overwritten by the applications in run-time. Installing a new RPM will also overwrite the previous file.

For more information about these settings, see comments in respectively file.

MGU Software

This chapter describes the MGU software to get a brief understanding of the main applications and where different functions are handled in the SW.

General

The software consists of the following parts:

- Boot loader
- Linux operating system and root file system stored on-board in local (NAND) flash file system.
- Media Gateway applications. There are three server applications on MGU, each using SCTP protocol and a well-known port that an external MX-ONE Service Node may connect to and communicate with application through. These applications are named “Device Board Server” (eridbs), “Media Control Application” (erimca) and “Operation & Maintenance Application” (erioma) using SCTP ports 2816, 2818 and 2817, respectively.
- Media Gateway commands. Not for normal usage. There are commands mainly for manufacturing and development purposes, but also a few possible to use for fault isolation. These are further documented in the mgw man-page (e.g. enter “man mgw” when logged in on MGU).

Installation and Upgrade

MGU utilizes RPM package manager for software and firmware installation and upgrade. There is only one RPM for the whole MGU installation and this RPM contains all software, and MSP and FPGA FM (firmware) images previously used.

The MGU RPM is named in the format **mgw-X.Y.Z-1.ppc.rpm**, where X represents - a new generation software, Y represents - functional extension and Z represents - fault correction. An RPM name can also be named **mgw-X.Y.Z.B-1.ppc.rpm** to denote a test version for upcoming official release. The B version number will be increased for each test build releases. Such an RPM must be used for internal testing only, and must be more recent test versions are available or when an official release is published.

MGU provides support for downloading the RPM from a software server using HTTP protocol. Installation is normally ordered by MX-ONE Service Node through O&M signaling port, but can be done through MGU upgrade or by using Linux rpm commands. Installation of the RPM could involve reboot of MGU; for example, if boot loader or Linux has been changed.

Upgrading the RPM with Linux rpm command is not recommended.

Boot Loader

The boot loader is the bootstrapping process that starts operating systems (Linux) on the MGU. It handles:

- Basic initialization of the MGU board
- Chip Select setup
- SDRAM configuration

- Linux boot

The boot loader have also a Programmable Built-In Self-Test (PBIST) support.

The bootloader also share configuration data with the OS (Linux) which resides on a NOR-flash memory.

The flash contains:

- Boot configuration parameters
- Manufacturer data (MAC addresses, serial number, etc)
- Product information (ROF, index numbers and revision information)
- Configuration data (IP-network and linux configuration data)

Printout sample from nor flash:

```
DISP *ROF_num = ROF 137 6304/1
DISP *ROF_rev = R1A
DISP *ROF_ser = T01D896676
DISP *eth0_mac = 00:13:5E:F0:AD:C3
DISP *eth1_mac = 00:13:5E:F0:AD:C2
DISP *eth2_mac = 00:13:5E:F0:AD:F4
DISP *eth3_mac = 00:13:5E:F0:AD:F5
DISP nfsroot = /mgu_root
DISP lilo_arg = root=/dev/mtdblock1 rw rootfstype=yaffs noatime
DISP autoupdate = no
DISP eth0_ip = 10.105.68.57/24
DISP autoboot = yes
DISP nfsboot = no
DISP phy0_mode = AUTO
DISP lan_active = LAN0
DISP lan_primary = LAN0
DISP eth2_ip = 10.105.68.58/24
DISP phy1_mode = AUTO
DISP eth1_ip = 10.10.1.2/24
DISP def_route = 10.105.68.1
DISP def_route1 = 10.10.1.1
DISP eth3_ip = 10.10.1.3/24
```

Operating System and Root File System

The operating system and root file system is based on Wind River® Linux version 1.4.

Device Board Server

The device board server (DBS) subsystem in MGU hosts the ISDN signaling and Device Board interface functions.

The message passing on Service Node (SN) interface is carried out using Stream Control Transmission Protocol (SCTP) on port 2816.

SCTP has many features but mainly message integrity and safe data delivery is currently in use.

The messages from the SN use configured multiple number identities (in the message header) to address the functions.

The ISDN and Device Board functions are described in other parts of this document.

Media Control Application

The Media Control Application (MCA) is a service on the MGU which main function is to provide control of media related functions for MX-ONE Service Node on SCTP port 2818. In short, these functions include:

- Creating and managing VoIP (Voice media) and FoIP (T38).
- Creating and managing secure VoIP streams, using SRTP/SRTCP.
- Controlling auxiliary functions like DTMF signal detection and playing recorded voice announcements.
- Quality of Service (RTCP/RTCP-XR, VLAN and Diffserv).

Operation & Maintenance Application

The Operation and Maintenance Application (OMA) is a service on MGU that provides an interface for MX-ONE Service Node on SCTP port 2817 for the following functions:

- Selection of market and time zone.
- Setting and retrieving run-time parameters.
- Alarm configuration and reporting.
- Inventory information (SW, FW and HW revisions).
- Network configuration.
- Installation & Upgrade of SW and FW.
- Installation of Recorded Voice Announcements.
- Selection of TDM synchronization source.
- Common restart functions, such as MGU restart, reboot and shutdown.

Capacity and Limitations

Device Board Interface

Table 5.1: Capacity and limitations in Device Board Interface

Signaling Protocols	2 Mbit HDLC (long signal format, i.e. up to 300 bytes payload) 128 Kbit UART short format 128 Kbit UART long format UART protocol with slow signaling is not supported
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Digital Trunk Interface

Table 5.2: Capacity and limitations in Digital Trunk Interface

Number of PRIs	8 ISDN PRI:s. Each 30B+D or 23B+D
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Fax Relay T.38

Table 5.3: Fax support and settings. Note that settings are fixed and are not configurable.

Max T.38 sessions	128 simultaneous sessions if only running T.38 sessions in MSP.
Supported fax signals.	V.21, V.25 and V.8 (Preamble/flags, CED, Ans and ANSam)

Table 5.4: T.38 Fax Configuration data used in MGU (Sheet 1 of 2)

TCF Procedure:	Remote TCF, the TCF data is passed across the IP network in the same way as any other page data.
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Table 5.4: T.38 Fax Configuration data used in MGU (Continued) (Sheet 2 of 2)

Redundancy:	Allow ECM faxes in T.38.
No maximal speed limit negotiated:	No speed limit.
Redundancy count for T30 messages:	Total 7 counts.
Redundancy Count Page Data	Total 7 counts.
ECM faxes in T38:	ECM Allow.
T.38 Packet loss concealment:	No T.38 Packet loss concealment
Small ECM packet handling:	Enable small T4 ECM packet instead of waiting for complete HDLC ECM frame.

Keycode Receiver

Table 5.5: KR resources

KR latency	In serial connection KR adds a delay of about 65 ms per default, but can be tuned down to about 18 ms at the cost of performance (see below). In parallel connection KR adds no delay.
Max KR resources	400 simultaneous KR if DetectionDelay = 40 ms, and only running KR resources in MSP 114 simultaneous KR if DetectionDelay = 18 ms, and only running KR resources in MSP
Start KR session	Less than 15 ms
Stop KR session	Less than 10 ms

Keycode Sender

Table 5.6: KS resources

Max KS resources	400 simultaneous KS
DTMF/MFC duration	1-255 ms, and continuous (=0)
DTMF/MFC level	0 to -36dBm0

Tone Sender

Table 5.7: TS resources

Max TS resources	400 simultaneous TS.
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Recorded Voice Announcement

Table 5.8: Recorded Voice Announcement

RVA download time (unloaded MGU)	This time is dependent not only on MGU load, but also on network and web-server conditions.
RVA activation time (unloaded MGU)	To activate media files takes about 12s/Mbyte before they are active and can be used. For a maximum file size installation, installation time is about 6-7 minutes. During activation, RVA feature is disabled (RVA sessions will be rejected by MGU).
Max RVA sessions	195 simultaneous sessions if only running RVA in MSP
Supported file formats	- RIFF (little-endian) data, WAV audio, ITU-T G.711 A-law, mono 8000 Hz - RIFF (little-endian) data, WAV audio, ITU-T G.711 μ -law, mono 8000 Hz
Max number of RVA files	250
Max file size	60 minutes (approximately 30Mbyte)
Start RVA session	To allocate and start a media player (RVA) session takes less than 50 ms
Stop RVA session	To stop a media player session takes less than 50 ms

Muti Party

Table 5.9: MP resources (Sheet 1 of 2)

Max MP resources	64 simultaneous MP resources
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Table 5.9: MP resources (Continued) (Sheet 2 of 2)

Max participants per MP resource	16 (Note: The MX-ONE Service Node limits the number of participants to 8)
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Voice Over IP

NOTE: Only symmetric Voice traffic are supported. Receiving and sending channel must use the same type of codec and packetization interval.

Table 5.10: Supported codecs and packetization intervals

Codec	Packetization intervals
G.711 a-law/ μ -law	5, 10, 20, 30, 40, 50 and 60 [ms]
G.729a	10, 20, 30, 40, 50, 60, 70 and 80 [ms]
G.729ab	10, 20, 30, 40, 50, 60, 70 and 80 [ms]
Clear channel	5, 10, 20, 30, 40, 50 and 60 [ms]

RTP/RTCP port range	Configurable from MX-ONE Service Node. Default range if not set is 50000..57768
Crypto Suites (SRTP)	The following encryption / authentication combos are supported: <ol style="list-style-type: none"> 1. AES-128 / HMAC-SHA1-80 2. AES-128 / HMAC-SHA1-32 3. AES-128 4. AES-192 / HMAC-SHA1-80 5. AES-192 / HMAC-SHA1-32 6. AES-256 / HMAC-SHA1-80 7. AES-256 / HMAC-SHA1-32
Fax PassThrough	Switching on detection of CNG, CED, V.21 flags, ANSam ANS/, COT V8bis, V22 and Bell 103 tones.

Max RTP sessions	Up to 290 sessions if only RTP is running in MSP, but depends heavily on settings and speech characteristics. NOTE: With “clear channel” max RTP sessions could be up to 404, but lower with other codecs. <i>See table 24 below for a few examples.</i>
Time to establish new RTP session	To allocate and start a new RTP session takes less than 30 ms
Time to close RTP session	To stop and free RTP session takes less than 20 ms

Table 5.11:Max RTP sessions for various settings STD = Standard EC with default settings, DFE = Dual Filter EC with default settings (EC type and settings is configurable through TDM MO)

Codec	Echo Cancellor	Crypto suite (see table 23)	Packetization Interval		
			10 ms	20 ms	30 ms
G.711	STD	-	140	240	290
G.711	DFE	-	140	200	230
G.711	STD	6	72	122	150
G.711	DFE	6	66	108	136
G.729ab	STD	-	114	134	154
G.729ab	DFE	-	114	110	124
G.729ab	STD	6	50	82	106
G.729ab	DFE	6	52	78	100

NOTE: In density figures above it has not been included the effect of using “Silence Suppression”, which when enabled might lower DSP load and thus increasing channel density.

In table below, the latency on VoIP channels is stated for different codecs and packetization intervals. Latency includes the path from TDM to VoIP encoding to packet network to decoding to TDM, i.e. GW to GW VoIP delay. Latency has been measured with fixed jitter buffer of same size as packetization interval. These are minimum latencies to be expected in GW to GW calls, and external packet network and endpoints might increase latency. Also use of encryption increases latency slightly.

Table 5.12:End to End VoIP delays with fixed jitter buffer equals packetization interval (Sheet 1 of 2)

Codec	Packetization Interval			
	5 ms	10 ms	20 ms	30 ms
G.711	25	35	45	Not measured

Table 5.12:End to End VoIP delays with fixed jitter buffer equals packetization interval (Continued) (Sheet

Codec	Packetization Interval			
	5 ms	10 ms	20 ms	30 ms
G.729ab	N/A	58	58	Not measured

IPsec Standards

Table 5.13:IPsec

Protocols	IKEv1, AH, ESP, IP Compression
CBC ciphers	AES-128 (default), AES-256, DES, 3DES, Blowfish, Twofish, Serpent
Digests	SHA1 (default), SHA2 (SHA-256, SHA-384, SHA-512), MD5
Diffie Hellman groups	1, 2 (default), 5
IP Compression	Deflate

Network Redundancy

Table 5.14:Network Redundancy fail over-time

Fail over time (Subnet Redundancy)	Fail over time will be approximately 8 seconds
Fail over time (Switched redundancy)	Fail over time is about 1 second when the active link fails. Fail over occurs also if the subnet's gateway is not reachable and no control traffic packets are received on the LAN port. The fail over time in these case will be approximately 8 seconds.

Factory Default Parameters

Table 5.15: Factory default parameters

Parameter	Value
eth0_ip	192.168.1.2/24
eth1_ip	192.168.2.2/24
eth2_ip	192.168.1.3/24
eth3_ip	192.168.2.3/24
default_route	192.168.1.1
default_route1	192.168.2.1

Speech Switching Time in case of GARP or NA

The following table indicates about the switch time of ongoing speech to another MAC-address when a Gratuitous ARP (Address Resolution Protocol) and/or unsolicited Neighbor Advertisement is received. The time also depends on the current load of the MGU.

Table 5.16: Speech Switching Time

No Calls	Switch Time (ms)
1	20
2	20
5	30
10	40
20	60
40	110
60	150
80	210
100	290

