Multi-service Access Gateway Implementation Agreement (UK Market based on ETSI H.248 profile)

MSF-IA-MEGACO.010-FINAL

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Abstract: This IA profile is for a multi-service Access Gateway (AG) that is suitable for deployment as an integral part an UK operators Next Generation Network (NGN). The AG provides support for two wire POTS/ISDN-BA and ISDN-PRA customer interfaces. This IA describes the overall network architecture, the behaviour of the AG and associated Call Agent and provides profiles for H.248 and Sigtran (IUA) interfaces.

This contribution updates/replaces the previous GMI2004 IA for Megaco/H.248 (document: MSF-IA-MEGACO.005-FINAL)

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To date the MSF has defined a number of detailed Implementation Agreements and detailed Test Plans for the signaling protocols between network components and is developing additional Implementation Agreements and Test Plans addressing some of the other technical issues such as QoS and Security to assist vendors and operators in deploying interoperable solutions.

The MSF welcomes feedback and comment and would encourage interested parties to get involved in this work program. Information about the MSF and membership options can be found on the MSF website http://www.msforum.org/

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MultiService Forum

The goal of the MSF is to promote multi-vendor interoperability as part of a drive to accelerate the deployment of next generation networks. To this end the MSF looks to adopt pragmatic solutions in order to maximize the chances for early interoperability in real world networks.

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In 2004, the MSF held its second "Global MSF Interoperability 2004" (GMI 2004) event that tested interoperability between next generation network elements situated in Asia, Europe and North America. GMI 2004 validated the MSF release 2 architectural framework and Implementation Agreements by subjecting them to interoperability testing based on realistic network scenarios.

Following the success of GMI 2004 the MSF work program continues to address the key technical barriers to next generation network deployments. Global MSF Interoperability 2006 (GMI 2006) will demonstrate a deployable and operationally ready IP telephony network with Network Management, enhanced Quality-of-Service (QoS) and security features. GMI 2006 will also demonstrate a service layer with application server, media server, and service broker functionality. This will enable the MSF to demonstrate a full end- to- end customer ready deployable network.

It is envisaged that GMI 2006 will provide an industry showcase that will:

- Assist carriers achieve their goal: to deploy flexible, best of breed products.
- Assist vendors achieve their goal: to market products more cost effectively.
- Display the global interoperability of the MSF architecture as referenced in the Release 2 architecture document.
- Demonstrate a network scenario that can be managed to specific quality standards.

The MSF welcomes feedback and comment and would encourage interested parties to get involved in this work program. Information about the MSF and membership options can be found on the MSF website http://www.msforum.org/

I. Applicability and Scope

This Implementation Agreement was created to describe the interface between a Call Agent and Access Gateway when using Megaco as a signaling protocol in preparation for GMI 2006 in a VoIP Network. Additional changes have subsequently been made based on operational experience in the GMI event.

This document updates the specification Implementation Agreement for MEGACO/H.248 Profile for a Media Gateway Controller/Access Gateway (MSF-IA-MEGACO.005-FINAL). This document updates MSF-IA-MEGACO.005-FINAL (written sometime ago) and incorporates changes to H.248 for POTS control with reference to the ETSI stimulus analogue line package, superceding the previous IA that supported the MSF specific packages defined for the GMI2004 event.

Figure 1 defines the reference architecture for the AG. It also illustrates how the AG fits into the overall NGN. The AG provides support for three different types of customer interfaces (POTS (sub-divided into DEL, E-PBX, L-PBX and DDI PBX), ISDN-BA, and ISDN-PRA) and a network interface for interconnecting into the IP backbone. The AG is under the control of the Call Agent (MGC). It should be noted that the entities within the AG are functional entities that are used to illustrate the user port types supported by the AG. They do not constrain any physical implementation of the AG.

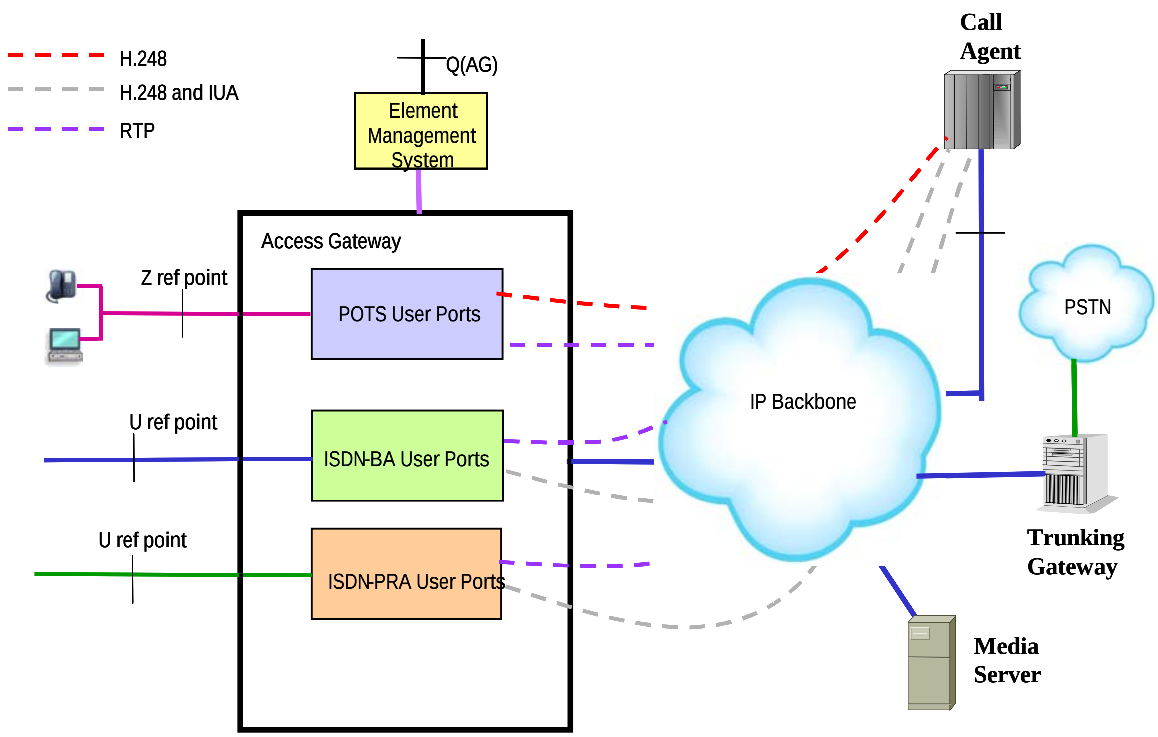

Figure 1: AG reference architecture

The AG supports the following general architectural requirements:

1. The AG shall be capable of being modeled as either single gateway or one or more Virtual Gateways.
2. The AG is not required to perform Local Switching (except under isolation/stand alone working).

3. Isolation/standalone working may be supported, but this is outside the scope of the IA.
4. Line test is performed via the QAG interface and is therefore considered to be outside the scope of the IA.
5. IP version 4 shall be used as the layer 3 protocol for signalling and media flows.
6. The AG shall provide support for the following alert/call progress tones:

    a. Dial Tone
    b. Special Dial Tone
    c. Message Waiting Tone
    d. Equipment Engaged Tone
    e. Number Unobtainable
    f. Busy tone.
    g. Howler
    h. Ringing Tone
    i. PSTN Call Waiting Tone
    j. Centrex Call Waiting Tone

7. The AG may optionally provide support for the following fixed end of call announcements:

    a. Unrecognised number
    b. Fault
    c. Number temporarily out of order
    d. No reply
    e. All lines busy
    f. Number unavailable
    g. Other party cleared
    h. Call cannot be connected
    i. Call Gapping
    j. Incoming Calls Barred
    k. No digits
    l. Service Termination

8. The AG shall provide support for generation of FSK signals for services such as caller number display.
9. The AG shall provide support for DTMF receivers.
10. The AG shall support loop disconnect dialling.
11. The AG shall support automatic metering of SPM equipment in order to avoid unnecessary overloading of the H.248 signalling link and Call Agent.
12. The AG shall provide support for a G.711 A law Voice codec and optionally for any other low bit rate Voice codec (e.g. G.729 A/B).
13. The AG shall support a Packetisation delay period in the range 5ms to 30ms in steps of at most 5ms.
14. Near end echo cancellation shall be supported.
15. The near end echo canceller shall be autonomously disabled by the AG upon detection of a fax/modem call as defined in G.168.
16. The near end echo canceller shall be disabled by the AG for ISDN-BA/ISDN-PRA lines under the instructions of the Call Agent, for example when the Call Agent determines that an ISDN-Bearer service of 64kbps unrestricted digital information has been requested.
17. Mapping of ISDN bearer services of 64kbps UDI and 7khz to a RTP payload format specifying G.711 A-law shall be supported. Mapping of these bearer services to “clear mode” (RFC 4040) may also be supported.
18. The AG shall support Dial Plans that are downloaded via H.248. The AG shall support at least one global public dial plan per AG and at least one private dial plan per user port for supporting services such as Centrex. The AG shall be capable of supporting a dial plan with a digit string of at least 100 characters.
19. Fax/Modem Relay, T.38, V.150, TTY relay are not required to be supported by the AG.
20. The AG shall support a timing reference that is network derived and thus synchronised to the network.
21. Support of a real time clock so that events that are reported to the Call Agent can be time stamped to a granularity of one hundredth of a second.

22. The AG shall have the capability to transport DTMF digits using ref [1] when non transparent codecs are used.

The AG shall not be required to create contexts containing more than two terminations.

I.A. *Customer Interfaces Supported*

I.A.1. POTS

The POTS interfaces supported by the AG are

- Direct Exchange Line (DEL) – used for supporting lines that have residential CPE or small call routing apparatus such as turn-key systems attached to them (see BS 6305)
- Loop Calling PBX (L/PBX) – for supporting PBX lines that use loop start protocol for initiating a call (see BS 6450)
- Earth Calling PBX (E/PBX) - for supporting PBX lines that use a ground start protocol for initiating a call (see BS 6450), and
- DDI PBX – for supporting analogue direct dial in PBX lines (see PD 7003)

I.A.2. ISDN-BA

The ISDN BA line card shall provide support for one or more basic rate ISDN interfaces and in particular it provides the ISDN “U” interface that is normally attached to an external NTE device

ISDN-BA services supported shall include

- ISDN D-channel packet (support of p-type and f-type frames)
- NMDS – See EN 301 141-1
- ISDN Call Control Signalling (support of s-type frames)

I.A.3. ISDN-PRA

The ISDNPRA line card shall provide support for one or more ISDN primary rate access interfaces over an E1 link. The line card shall provide the “U” interface for attaching to an external NTE device.

II. SIGTRAN – ISDN Q.921 User Adaptation Layer IUA)

ISDN User Adaptation (IUA [10] shall be used for transporting the D-channel (p-type, f-type and s-type frames) information to a remote entity e.g. Call Agent/Packet handler. The protocol model in ref [10] defines the functional entities Signalling Gateway (SG) and Application Server Process (ASP). The SG is the entity where the ISDN D-channel is terminated and then transported using the "IUA" protocol to an ASP. The AG performs the functions of the SG and the Call Agent performs the functions of the ASP.

II.A. ISDN-BA lines

It should be noted that H.248 is required to be supported in conjunction with IUA in order to provide a packetisation version of the ISDN service. In particular H.248 shall be used for the handling the adaptation of the B channels to RTP media streams, applying tones and DTMF digit collection (where applicable).

The LAP-D state machine (including TEI assignment and management procedures) shall reside within the AG and shall conform to ETS 300 402-1. Automatic TEI values in the range 64-126 may only be requested by the TE. Non-automatic TEI values in the range 0-63 shall be created in the assigned state by the AG upon activation of the layer 1 and these values can then be freely used by either the TE or the Call Agent.

"Point to point" or "point to multi-point" procedures are solely under the control of the Call Agent and have no impact on the AG.

The AG shall support either Permanent activation of the layer 1 or activation of layer 1 on a per call basis and this mode shall be configurable via the QAG. These modes of operation shall be under the control of the AG.

Activation of Loop backs within the access digital section (e.g. loopback at the NT1) shall be under the control of the AG. When a loopback is applied it will also be necessary for the AG to inform the Call Agent that the ISDN access is unavailable for the presentation of incoming calls. This can be achieved by using the H.248 Service Change mechanism.

II.B. ISDN-PRA lines

The AG requirements are as per the ISDN-BA section.

It should be noted that Time slot "0" is terminated in the AG and therefore any OAM procedures related to the access digital section (e.g. Loopback at the NT1) are handled entirely within the AG.

III. MEGACO Protocol Behavior

III.A. Device Control Protocol

H.248 version 2 as defined by ITU-T Recommendation H.248.1 (05/2002) shall be used with text based encoding. The H.248 protocol version “2” shall be included by the AG in the “ServiceChangeVersion” parameter to facilitate future upgrades to higher protocol versions of H.248. In addition the AG shall include the H.248 protocol version in the H.248 message header.

III.B. Profile

The H.248 profile specified by this IA is based on the UK ETSI based H.248 profile and shall be named as “ETSI_ARGW” version “1”. The profile and its version number shall be included by the AG in the “ServiceChangeProfile” parameter with the format “ETSI_ARGW/1”.

III.C. MEGACO Commands

Based on [2] the following commands shall be supported.

MEGACO Command	**MEGACO Parameters**	**Notes**
Add	TerminationId M: MediaDescriptor SG:SignalDescriptor E:EventDescriptor	
Modify	TerminationId M: MediaDescriptor SG:SignalDescriptor E:EventDescriptor DM: DigitMapDescriptor	
Subtract	TerminationId Auditdescriptor	Statistics may be returned in the response.
Audit Value	TerminationId Auditdescriptor	Context and Termination id wildcarding supported
ServiceChange	TerminationId ServiceChangeDescriptor	
Notify	TerminationId ObservedEventsDescriptor ErrorDescriptor	

III.D. Megaco Packages

This section lists the required packages for the AG and the status for each of the packages.

Package ID / Name	Defined In	Status
g / GENERIC	H248.1	Mandatory
root / BASE ROOT	H248.1	Mandatory
nt / NETWORK	H248.1	Mandatory
xdd / EXTENDED DTMF DETECTION	H248.16	▪ Mandatory for POTS and ISDN ▪ Not applicable for DDI
tdmc / TDM CIRCUIT	H248.1	Mandatory
alert / ENHANCED ALERTING	H248.23	▪ Mandatory for POTS ▪ Not applicable for DDI and ISDN
andisp / ANALOGUE DISPLAY	H248.23	▪ Mandatory for POTS ▪ Not applicable for ISDN
rtp / RTP	H248.1	Mandatory
ocp /Overload Control Package	H248.11	Mandatory
it / Inactivity Timer	H248.14	Mandatory for UDP transport.
qac / Quality Alert Ceasing	H248.13	Mandatory
stimal / STIMULUS ANALOGUE LINE	H248.34	▪ Mandatory for POTS and DDI ▪ Not applicable for ISDN
amet / AUTOMATIC METERING	H248.26	▪ Mandatory for POTS ▪ Not applicable for ISDN and DDI
cg / CALL PROGRESS TONES GENERATION	H248.1	Mandatory for POTS and ISDN
an / GENERIC ANNOUNCEMENT	H248.7	Mandatory for POTS and ISDN
xcg / EXPANDED CALL PROGRESS TONES GENERATION	Q1950	Mandatory for POTS and ISDN
srvtn/ BASIC SERVICES TONE GENERATOR	Q1950	Mandatory

III.D.1. User Agent Signals and Events Requirements

The following attributes shall be supported by the Access Gateway

III.D.1.a) Generic (g)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Events	Signal Completion (sc)	M	
	Cause (cause)	M	If requested by the MGC, shall be supported.

III.D.1.b) Base Root Package(root)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Properties	MaxNumberOfContexts	O	
	maxTerminationsPerContext	O	
	normalMGCExecutionTime	O	
	normalMGExecutionTime	O	
	MgprovisionalResponseTimerValue	O	
	MGCprovisionalResponseTimerValue	O	
	MGCOriginatedPendingLimit	O	
	MGOriginatedPendingLimit	O	

III.D.1.c) Network(nt)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Properties	Maximum Jitter buffer(jit)	O	
Events	Network failure (netfail)	O	
	Quality alert(qualert)	O	
Statistics	Duration(dur)	M	
	Octets sent(os)	M	
	Octets received(or)	M	

III.D.1.d) Extended DTMF detection package(xdd)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Events	DigitMap Completion event(xce)	M	
	DTMF digits (d0-do)	M	

III.D.1.e) TDM Circuit package(tdmc)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Properties	Echo cancellation(ec)	M	
	Gain Control(gain)	O	May be configurable via management.

III.D.1.f) Enhanced Alerting(alert)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Signals	Ringing signal (ri)	M	
	Ring splash (rs)	M	
	Call waiting (cw)	M	

III.D.1.g) Analogue Display(andisp)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Signals	Display with alerting (dwa)	M	
	Display Data (data)	M	

III.D.1.h) RTP Package(rtp)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Statistics	Packets sent (ps)	M	
	Packets received (pr)	M	
	Packet loss (pl)	M	
	Jitter (jit)	M	
	delay	M	
Events	pltrans	O	

III.D.1.i) Overload Control Package(ocp)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Events	mg_overload	M	

III.D.1.j) Inactivity Timer Package(it)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Events	it/ito	M	

III.D.1.k) Quality Alert Ceasing(qac)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Events	qac	M	

III.D.1.l) Stimulus Analogue Line (stimal)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Signals	pulsedsig – pulsedNoBattery, pulsedReducedBattery	M	Pulsed no battery results in the AGW applying an “Disconnect Clear”. Pulsed reduced battery results in the AGW applying an “end of call” signal. The duration of these signals is configurable within the AGW on a per port basis.
	stedsig – normalPolarity, reversePolarity, reducedBattery	M	Normal Polarity results in the AGW applying “Normal Line Feed”. Reverse Polarity results in the AGW applying “Reverse Line Feed”. Reduced Battery results in the AGW applying “Parked Feed”.
	Callfinished	M	Results in the AGW inserting the analogue port into the free/idle condition.
	Digits	M	DDI PBX digits

Events	stedsig – offHook, onHook, b-wireConnectedtoearth	M	Off Hook indicates a seize condition On Hook indicates a clear condition B-wire connected to earth indicates that the EC-PBX has not removed the earth from the b-wire during the seize validation timer.
	pulsedsig – registerrecall	M	Register Recall signal detected by the AGW. The register recall signal is electrically equivalent to a Loop Disconnect (LD) Digit "1". See section 8.1.10 on the detailed procedures used by the AGW for distinguishing between a Digit "1" and a "Recall" signal.
	onHook	M	DDI exchange release signal

III.D.1.m) Automatic Metering (amet)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Signals	Enable metering (em)	M	
	Meter pulse burst (mpb)	M	
	Phased metering (phsm)	O	
Events	Periodic report (pr)	M	

III.D.1.n) Call Progress tones generator package(cg)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Signals	dt	M	Dial Tone.
	rt	M	Ringing Tone.
	bt	M	Busy Tone.
	ct	M	Equipment Engaged Tone.

III.D.1.o) Generic Announcement(an)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Signals	apf/an = unrecnuman	M	Unrecognised Number.
	apf/an = fltan	M	Fault.
	apf/an = numtooan	M	Number temporarily out of order.
	apf/an = noreplyan	M	No Reply
	apf/an= linesbusyan	M	All lines Busy.
	apf/an = nuan	M	Number Unavailable.
	apf/an = opcan	M	Other Party Cleared.
	apf/an =callnotconan	M	Call cannot be connected.
	apf/an = callgapan	M	Call Gapping .
	apf/an = icban	M	Incoming Calls Barred.
	apf/an = nodigitsan	M	No Digits
	apf/an = servterman	M	Service Termination

III.D.1.p) Expanded Call Progress tones generator package(xcg)

Attribute	Subfield	Mandatory/ Optional/	Remarks

		Not Required	
Signals	vac	M	Number Unobtainable Tone.
	roh	M	Howler (Off-Hook Warning) Tone .
	spec	M	Special Dial Tone.

III.D.1.q) **Basic Service Tone Generator(srvtn)**

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Signals	mwt	M	Message Waiting Dial Tone.

III.E. Termination Naming Conventions

III.E.1. Physical Terminations

Analog POTS lines and ISDN-BA/ISDN-PRA "B" channel terminations shall be uniquely identified. The naming scheme/structure shall be configurable in the AG and the Call Agent, as it is network operator dependent. The text below gives some informative guidelines for such a naming scheme/structure.

For example the naming convention for a POTS termination could be of the form **aln/mmm/nnn** where the prefix "aln/" is a fixed value, and where "mmm" and "nnn" represent non-zero integer values identifying the AG analog port number. An analog port number 6 on line card 1 of an AG could be referred to by the name **aln/1/6**.

For example the naming convention for a Basic Rate ISDN "B" channel termination could be of the form **ba/ppp/rrr/sss**, where the prefix "ba/" is a fixed value, "ppp" and "rrr" represent non-zero integer values identifying the ISDN-BA port number and "sss" represents an integer of value 1 or 2 to identify the B1 and B2 channel terminations respectively. A B1 channel of ISDN-BA port 3 on line card 5 would be referred to by the name **ba/5/3/1**.

For example the naming convention for a Primary Rate ISDN "B" channel termination could be of the form **pra/ppp/rrr/sss**, where the prefix "pra/" is a fixed value, "ppp" and "rrr" represent non-zero integer values identifying the ISDN-PRA port number and "sss" represents an integer of value 1 or 30 to identify one of the 30"B" channels. A B30 channel of ISDN-PRA port 1 on line card 10 would be referred to by the name **pra/10/1/30**.

III.E.2. RTP Terminations

RTP terminations are ephemeral terminations and are created by an ADD command and destroyed by a SUBTRACT command. They only exist for the duration of their use and the AG may use any naming convention to uniquely identify the RTP terminations.

III.F. Topology Descriptor

The topology descriptor may be supported by an AG.

III.G. Service Change Descriptor

The Gateway shall allow one primary and one or more secondary MGCs to be provisioned for registration.

The MGC SHALL be able to control multiple AGWs simultaneously. Support of 1 or more virtual AGWs as defined in H.248.1 Section 11.1 is optional.

III.H. Heartbeat

When UDP is used as the transport protocol, then the Call Agent shall use the “AuditValue” command against the “ROOT” termination with an empty “Audit Descriptor” as a keep alive message for detecting loss of communications with the AG in the absence of other H.248 messages. If no reply is received to the “AuditValue” message, then the Call Agent shall determine that it has lost communication with the AG. It should be noted that this keep alive mechanism also complements the H.248.14 “Inactivity Timer” package [6] used by the AG for detecting loss of communications with the Call Agent.
Loss of communication can be due to:

- Loss of communication path between the MGC and AGW
- MGC becoming Out Of Service

This will be the only keep alive mechanism used. On the inactivity timer expiring, on the MG, the following actions shall be performed:

- AGW shall send Notify on Root, with event=ito to the primary MGC.
- If a Reply to the Notify is received, than the AGW will restart the Inactivity timer and consider the MGC still accessible.
- If no Reply to the Notify is received then the AGW will send ServiceChange=Root with Method=disconnect, Reason=”MGC impending failure” to the primary MGC.
- If a Reply to ServiceChange is received (without ServiceChangeMgcId of new MGC), than the AGW will restart the Inactivity timer and consider the MGC still accessible.
- If no Reply is received then the AGW will send ServiceChange=Root with Method=disconnect to the secondary MGC (if configured on the MG).
- The AGW will periodically send . ServiceChange=Root with Method=disconnect to the primary and secondary MGC until it receives a Reply with ServiceChange=Root.

Note if necessary once the communication to MGC is restored , the MGC can audit the status of each calls and retrieve information on all active contexts by auditing each termination.

III.I. Null Transaction Id

All Transaction Replies shall have transaction id with the following exception:

- If the receiver cannot determine a valid transaction id, than it will send with null transaction id and a single error descriptor 403. Please refer to section 8.2.2 of [2].

III.J. Transaction Timers

Gateways and MGCs SHALL keep application level transaction timers as outlined in Section 8 of ITU H.248.1

III.K. Transport

H.248 messages shall be transported using UDP/ALF.

III.L. Encoding

Conforming Gateways SHALL support text encoding.

III.M. Timestamp

The AG shall include a time stamp expressed in UTC in every Service Change and NOTIFY commands.

IV. Voice Codecs

For this IA, support for following codecs is mandatory :-

- G.711 (A-law or mu-law)
- Clearmode for ISDN in the instance of fallback.

Support for additional codecs (e.g. G.729 A/B) is optional.

The Access Gateway must support the packetization periods of 10, 20 or 30ms, except where the packet size is explicitly defined by the codec, e.g. G.723 (30ms). The packetization period may vary for each codec, for example G.726-32 could use 20ms and G.711 could use 10ms. Support for other packetization periods is optional.

V. Echo Cancellation

Access Gateways shall support echo cancellation in the receive direction (i.e. canceling echoes generated at the Gateway’s end of the call). This is also known as near end echo cancellation. Typical lengths for the echo tail required are dependent on the subsequent path taken through the PSTN but a figure of 128mS is reasonable to cover all occurences.
The Access Gateway shall,

- Autonomously enable the near end echo cancellation when a G.711 A law codec without silence suppression is specified.
- Autonomously disable the near end echo canceller (if already enabled) upon detection of fax/modem call as defined in G.168
- Autonomously disable the near end echo cancellation when a “clearmode” [14] codec is specified.

VI. Modem, Fax and TTY Support

Access Gateways shall detect the presence of voiceband data, i.e. modem or fax. This is detected by monitoring for the presence of the 2100 Hz tone defined in V.25 and the modified, amplitude modulated 2100 Hz tone defined in V.8. Access Gateways are required to detect the 2100 Hz tone with phase reversal as per section 4.1 of G.168 to ensure that echo cancellation and silence suppression is turned off.

In addition, Gateways shall be able to detect fax/modem tones both on their circuit side and packet side.

If G.711 is not included as a supported codec in the SDP then autonomous switchover does not occur.

If fallback to G.711 occurs, the Call Agent is responsible for ensuring that G.711 codec is used end-end for the call.

VII. DTMF Digits and Telephony Tones

DTMF in-band digit support is out of scope,this section is optional.

If supported, DTMF digits shall be handled as specified in section 9 of [18].

In summary this specifies

- Access Gateways shall have the capability to transport DTMF digits using [1] or in-band in the codec or both.
- Media Gateways shall also support notification of DTMF digits or tones via MEGACO Notify commands. Digits are reported only to the Call Agent and are not sent in the RTP stream.
- Codec negotiation is used to determine whether DTMF digits should be transported via [1] or in-band. Support for 1] is specified using the RTP payload format ‘telephone-event’. If this payload format is specified in the SDP descriptor then [1] should be used if supported, otherwise in-band transport is used.

VIII. Security Aspects

VIII.A. NAT and Firewall traversal

Not applicable. Access Gateways are trusted elements within the Operator’s network.

VIII.B. Encryption

A private IP network is used to carry the signaling and media traffic from an AG. Therefore there is no real need to provide any additional security mechanisms within the private IP network. But as an option an AG may support authentication using for example IPSEC AH.

IX. Redundancy and resilience

This section describes the redundancy and resilience requirements.

- The AG shall have better than 99.999% availability in order to achieve the same service level agreements as offered on current PSTN networks.
- The architecture of an AG shall be redundant such that no single point of failure can deprive all that AG's customers of service.
- The AG shall support a primary network link and at least one secondary link to ensure that no loss of service occurs under primary link failure. When failure does occur then the AG shall switch over to one of the protected links. The switch over shall occur within tens of milliseconds in order to minimum disruption of media and signaling traffic.
- Under fault conditions the AG shall be capable of re-registering with a secondary “Call Agent”.
    - The AG shall support a list of secondary Call Agents with a minimum of 1 secondary Call Agent within the list.
    - The change over mechanism for POTS terminations and ISDN-B channels shall be as defined by H.248.1 clause 11.5.For ISDN-D channels the change over shall be as defined in ref [10] in particular by utilizing the ASP messages.
    - The AG shall support revertive switch over back to the repaired Call Agent. This mode of operation shall be triggered by the current working Call Agent by using the Hand off procedures described in H.248.1 clause 11.5. For ISDN D Channels the revertive mode of operation shall be triggered by the repaired Call Agent, by using the override mechanism described in ref [10] clause 5.2.2.

  - Under change over the AG and Call Agent must preserve at least those calls that are in a stable and established state.

- The AG shall preserve stable and established calls under restart (e.g. warm start-up due to processor failure) conditions.

IX.A. Timing Considerations

The AGW shall support a timing reference that is network derived and thus synchronized to the network.
Support of a real time clock so that events that are reported to the Call Agent can be time stamped to a granularity of one hundredth of a second.

X. Management Information Model

The AG is a device that is managed through the QAG interface and there are a number of attributes that are required to managed in the AG in order to provide a end to end VoIP service.

This section defines a management information model which will used as the framework for defining all the objects and attributes that are required to be managed in the AG device. The management information model is described using the Unified Modeling Language (UML) syntax. The management information model is management protocol agnostic, since the exact management protocol to be used for managing the AG is outside the scope of this IA.

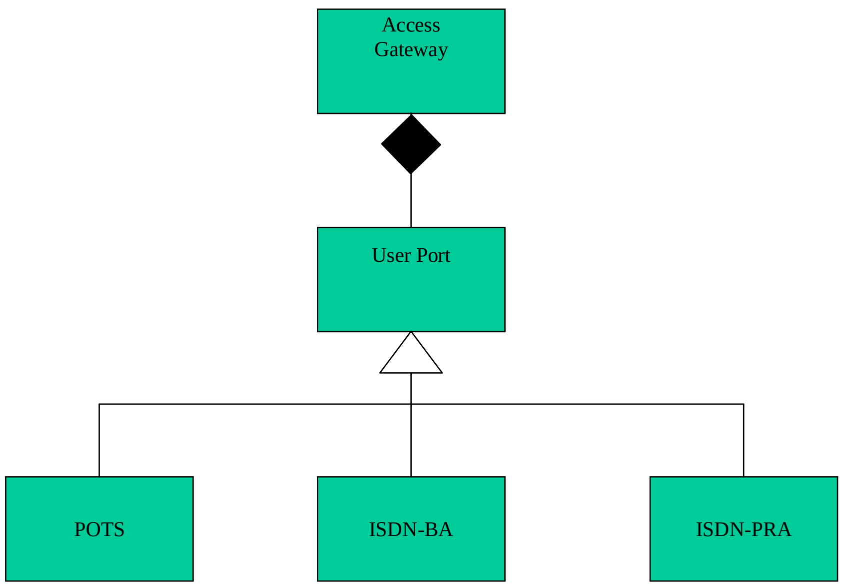

Figure 2: AG management information Model

The model reads as; An AG has one or more network interfaces and the AG contains one or more UserPorts. Three types of user ports exist and these include POTS and ISDN-BA and ISDN-PRA. It should be noted that by using inheritance the model allows other user port types to be added easily.

X.A. Description of Classes and Attributes

This section provides a textual description of all the classes contained in the Management Information model and the attributes associated with each of the classes.

Access-Gateway – This class represents the AG entity.		
Attribute	Values	Description
Vendorname	Text String	The name of the vendor that supplied the AG device
DeviceType	Text String	Vendors model number for the AG device
HardwareVersion	Text String	The hardware revision for the AG device
SoftwareVersion	Text String	The software revision for the AG device
AdminStatus	Unlocked,locked, shuttingdown, testing	Indicates the desired state of the AG device
AG Address	Choice of IPv4 Address or Domain name	Address of the AG device – Addresses shall be encoded as defined in H.248.1.
Primary CA address	Choice of IPv4 Address or Domain name plus a port number	The address of the primary CA that the AG communicates with. Addresses shall be encoded as defined in H.248.1 The Port number with a default value of 2944
Secondary CA addresses	Sequence of {Choice of IPv4 Address or Domain name plus a port number}	A list of secondary CA addresses in descending priority order – these secondary CA addresses are used when failure of the primary CA occurs. Addresses shall be encoded as defined in H.248.1. The Port number with a default value of 2944
Start Digit Timer	Integer	Timer to receipt of the first digit expressed in seconds
Long Digit Timer	Integer	Timer to receipt of a subsequent digit expressed in seconds, whilst a digit match has not occurred.
Short Digit Timer	Integer	Timer to receipt of a subsequent digit expressed in seconds when a match has occurred, but there are other candidate digit strings for match.
FSK prior to ringing timers	Sequence of Integers	The sequence of timers shall include silence timer 1, duration of DT-AS, Silence Timer 2, Channel Seizure Signal Duration, Mark Signal Duration and Silence Timer 3. These timers are expressed in milliseconds.
FSK during ringing timers	Sequence of Integers	The sequence of timers expressed in milliseconds shall include Silence Timer 4 and Silence Timer 5.
FSK during off-hook	Sequence of Integers	The sequence of timers expressed in milliseconds shall include Silence Timer 1, timer for receipt of TE-Ack and Silence Timer 2.
Equipment Engaged Tone Duration	Integer	Duration expressed in milliseconds
Number Unobtainable Tone Duration	Integer	Duration expressed in milliseconds
Busy Tone Duration	Integer	Duration expressed in milliseconds
Howler Tone Duration	Integer	Duration expressed in milliseconds
Ringing Tone Duration	Integer	Duration expressed in milliseconds
Call Waiting Tone PSTN Duration	Integer	Duration expressed in milliseconds
Call Waiting Tone Centrex	Integer	Duration expressed in milliseconds

Duration		
Unrecognised Number Announcement Cycles	Integer	Number of cycles of the number unrecognised announcement to be applied.
Fault Announcement Cycles	Integer	Number of cycles of the fault announcement to be applied.
Number temporarily out of order Announcement Cycles	Integer	Number of cycles of the temporarily out of order announcement to be applied.
No reply Announcement Cycles	Integer	Number of cycles of the no reply announcement to be applied.
All lines busy Announcement Cycles	Integer	Number of cycles of the all lines busy announcement to be applied.
Number unavailable Announcement Cycles	Integer	Number of cycles of the number unavailable announcement to be applied.
Other party cleared Announcement Cycles	Integer	Number of cycles of the other party cleared announcement to be applied.
Call cannot be connected Announcement Cycles	Integer	Number of cycles of the call cannot be connected announcement
Call Gapping Announcement Cycles	Integer	Number of cycles of the call gapping announcement to be applied.
Incoming calls barred Announcement Cycles	Integer	Number of cycles of the incoming calls barred announcement to be applied.

UserPort – This class represents the User Port entity and is the SuperClass		
Attribute	Values	Description
Portlabel	Text String	Label such as DN in order to identify the port.
H.248TerminationId	Text String	Label used by H.248 in order to address the User port

POTS – This class represents the POTS User Port entity and is a subclass of the UserPort.		
Attribute	Values	Description
POTS signaling Method	Direct exchange line, earth calling, loop calling, ddi)	Identifies the analogue port Type
Gain Type	Automatic gain, fixed	Automatic or fixed gain to be applied to the line.
Fixed Gain values	Transmit Gain, Receive Gain	The transmit and receive gain are expressed in Decibels with a negative value indicating a loss.
Disconnect Clear Duration	Integer	Duration expressed in milliseconds
End Of Call Clear Duration	Integer	Duration expressed in milliseconds
DDI Exchange Released Duration	Integer	Duration expressed in milliseconds
Register Recall Duration	Minimum and Maximum Duration	Minimum and Maximum duration of the Register Recall pulse expressed in milliseconds.
Ring Splash Duration	Integer	Duration expressed in milliseconds
SPM type	12khz or 16khz	The frequency of SPM pulses to be applied to the line.

ISDN-BA – This class represents the ISDN-BA User Port entity and is a subclass of the UserPort.

Attribute	Values	Description
S Type Frame supported	No, Yes	Support of ISDN Call Control frames
F Type Frame supported	No, Yes	Support of frame relay packets.
D Type Frame supported	No, Yes	Indicates support of D-channel packets
Layer 1 activation	Permanent, Per Call	Activation of ISDN Layer 1 either permanently or on a per call basis.

ISDN-PRA – This class represents the ISDN-PRA User Port entity and is a subclass of the UserPort.		
Attribute	Values	Description
None Currently Identified.		

XI. References

[1]	IETF – draft-ietf-avt-rfc2833bis-12.txt RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, November 2005
[2]	ITU-T Recommendation H.248.1 (05/2002) – Gateway control protocol: Version 2.
[3]	ITU-T Recommendation H.248.8 (03/2002) – Gateway control protocol: Error Code and Service Change Reason Description.
[4]	ITU-T Recommendation H.248.11 (11/2002) – Gateway control protocol: Media gateway overload control package.
[5]	ITU-T Recommendation H.248.13 (03/2002) – Gateway control protocol: Quality Alert Ceasing package
[6]	ITU-T Recommendation H.248.14 (03/2002) – Gateway control protocol: Inactivity timer package
[7]	ITU-T Recommendation H.248.16 (11/2002) – Gateway control protocol: Enhanced digit collection packages and procedures.
[8]	ITU-T Recommendation H.248.23 (7/2003) – Gateway control protocol: Enhanced Alerting packages
[9]	ITU-T Recommendation H.248.26 (7/2003) – Gateway control protocol: Enhanced analog lines packages
[10]	IETF – RFC 4233 Integrated Services Digital Network (ISDN) Q.921-User Adaptation Layer, January 2006
[11]	ETSI – ETS 300 402-1 Integrated Services Digital Network (ISDN), Diigtal Subscriber Signalling System Number 1 (DSS1) protocol; Data Link Layer: Part 1: General Aspects
[12]	ETSI – EN 301 141-1 Integrated Services Digital Network (ISDN), Narrowband Multi-Service Delivery System (NMDS); Part 1: NMDS Specification
[13]	MSF2003.073 MGCP IA Call Agent <-> User Agent
[14]	RFC 4040 April 2005 - RTP Payload Format for a 64 kbit/s Transparent Call
[15]	British Standards Institute - BS 6305 – 1992 Specification for General Requirements for Apparatus for Connection to Public Switched Telephone Networks run by certain Public Telecommunication Operators.
[16]	British Standards Institute - BS 6450 – Private Branch Exchanges for Connection to Public Switched Telephone Networks run by certain Public Telecommunication Operators.
[17]	British Standards Institute - PD 7003:1996 Essential requirements for terminal equipment intended for connection to analogue interfaces of the PSTN using direct dial in.
[18]	MSF2003.059 SDP usage and codec negotiation IA.
[19]	ITU-T Recommendation H.248.4 (2000), Gateway control protocol: Transport over Stream Control Transmission Protocol (SCTP).
[20]	ITU-T Q1950 Bearer Independent Call Control (BICC)

End of Implementation Agreement