H.248 Implementation Agreement Between a Call Agent and an IP Trunking Gateway

MSF-IA-MEGAC0.003.01-FINAL

MultiService Forum Implementation Agreement

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Abstract: This is the MSF Implementation Agreement for the H.248 interface between a Call Agent and Trunk media gateway. This contribution updates/replaces the previous GMI2002 Implementation agreement for Megaco/H248 for a Media Gateway Controller/Trunking gateway using IP trunks (document: MSF-IA-MEGACO.003-FINAL) and has been amended and updated to include interoperability issues seen at GMI 2004.
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 deployment in real world networks.

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 a "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.

Global MSF Interoperability 2004 (GMI 2004) demonstrated a deployable and operationally ready IP telephony network with Network Management, enhanced Quality-of-Service (QoS) and security features. GMI2004 also demonstrated a service layer with application server, media server, and service broker functionality.

GMI2004 provided an industry showcase that:

- Assisted carriers achieve their goal: to deploy flexible, best of breed products.
- Assisted vendors achieve their goal: to market products more cost effectively.
- Displayed the global interoperability of the MSF architecture as referenced in the Release 2 architecture document.
- Demonstrated a network scenario that was 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

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1 Applicability and Scope

This Implementation Agreement was created to describe the interface between a Call Agent and Trunking gateway when using Megaco as a signaling protocol in preparation for GMI 2004 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/Trunking Gateway using IP Trunks (MSF-IA-MEGACO.003-FINAL). The document MSF-IA-MEGACO.003-FINAL was written sometime ago and since than some of the drafts have expired and a subset has been incorporated in to ITU H248 series. In addition this document adds more details in terms of events/parameters for each package. Figure 1 shows the MSF architecture diagram for GMI 2004 and highlights the applicable MEGACO interfaces for this IA.

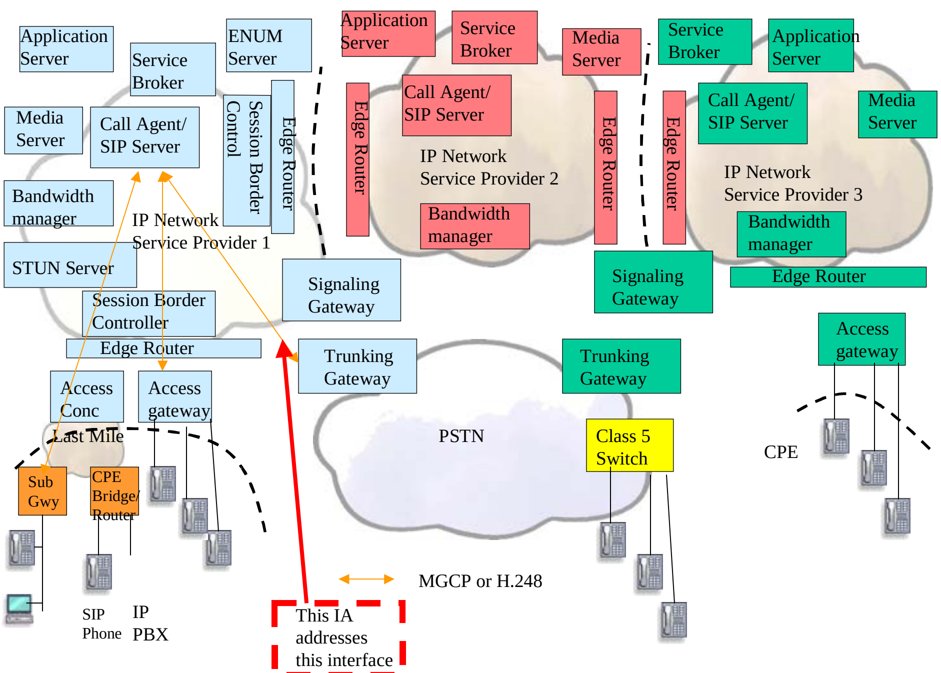


Figure 1: Architecture diagram indicating applicable interfaces for this IA in Service Provider 1

This IA addresses the interface between the Call Agent and the Trunking gateway. The Trunking gateway provides bearer interface between the Class 5/4 switch and the IP Network.

This IA will support the following types of interface to the Class5/4 switch

- T1/E1 ISUP trunk interface.

Trunking Gateways – these are devices resident at either Central Office or Remote Cabinet locations. These devices support TDM interfaces (T1/E1) for inter-connecting to the PSTN. The number of such interfaces may vary from card to card. Typically, multiple T1/E1s are supported on a single card. On the packet/ IP side, they are typically ethernet. They are trusted devices in the sense that they are owned by the N/W Operator and are thus are part of the Operator's private IP address space. Therefore, NAT & FW control are not appropriate for such devices

2 MEGACO Protocol Behavior

2.1 Device Control Protocol

MEGACO Version 2 shall be supported as specified in ITU document H248.1(05/2002). The H.248 profile specified by this IA shall be named as “MSF TGW” version “1”. The profile and its version number shall be included by the MG in the “ServiceChangeProfile” parameter with the format “MSF TGW/1”.

2.2 MEGACO Commands

Based on the H248.1 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	
Subtract	TerminationId	Statistics may be returned in the response.
Audit Value	TerminationId Auditdescriptor	Context and Termination id wildcarding supported
ServiceChange	TerminationId ServiceChangeDescriptor	
Notify	TerminationId ObservedEventsDescriptor	

2.3 Megaco Packages

For this profile the following mandatory and optional packages are required to be supported by both the Trunking gateway and the Call Agent.

Package Name / Id	Defined In	Status / Comments
Generic / g	[1]	Mandatory
Base Root / root	[1]	Mandatory
Basic Continuity Package / ct	[1]	Mandatory
Network package / nt	[1]	Mandatory
TDM Circuit package/tdmc	[1]	Mandatory
Fax,Text,Modem Detection Package/ ftmd	[2]	Optional
Call progress tone generator package / cg	[1]	Mandatory
Basic DTMF Generator package /	[1]	Mandatory..

dg		
DTMF detection package / dd	[1]	Mandatory.
Overload Control Package / ocp	[7]	Mandatory..
Inactivity Timer / it	[8]	Mandatory.
Call Type Discrimination package / ctyp	[2]	Mandatory. Note that fax/modem supported by G711 in the GMI2004 event (i.e. T38 optional) and thus the ftmd package could suffice.
RTP / rtp	[1]	Mandatory.
Quality Alert Ceasing / qac	[10]	Optional

2.3.1 User Agent Signals and Events Requirements

There are a set of events and signals that are defined for each package. For the purpose of a ISUP Trunking gateway, the following subsections define the mandatory event and signals that should be supported by the Trunking gateway for the set of mandatory packages.

2.3.1.1 Generic (g)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Events	Signal Completion (sc)	M	
	Cause (cause)	M	

2.3.1.2 Continuity (ct)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Events	Completion(cmp)	M	
Signals	Continuity test(ct)	M	
	Respond(rsp)	M	

2.3.1.3 Network(nt)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Properties	Maximum Jitter buffer(jit)	O	

Events	Network failure (netfail)	O	Note_1
	Quality alert(qualert)	O	
Statistics	Duration(dur)	M	
	Octets sent(os)	M	
	Octets received(or)	M	

Note_1: ITU H248.8 Error Code #: 530 (Temporary Network failure) and Error Code #: 531 (Permanent Network failure) can be used instead of the Eventid:netfail. In case autonomous failure discovery, the gateway can send ServiceChange message.

2.3.1.4 TDM Circuit package(tdmc)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Properties	Echo cancellation(ec)	M	
	Gain Control(gain)	M	

2.3.1.5 Basic DTMF Generator package(dg)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Signals	DTMF character 0-DTMF character D (d0-dd)	M	

2.3.1.6 DTMF detection package(dd)

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

2.3.1.7 Call Progress tones generator package(cg)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Signals	Dial tone (dt)	NR	Line side only
	Ringing Tone(rt)	M	

	Busy tone(bt)	O	
	Congestion Tone(ct)	O	
	Special information tone(sit)	NR	Line side only
	Warning Tone(wt)	O	
	Payphone recognition tone(prt)	NR	Line side only
	Call waiting Tone(cw)	NR	Line side only
	Caller waiting tone(cr)	NR	Line side only

Busy Tone/Cong Tone would not typically be connected via the TGWMG since the condition would be signalled in ISUP REL with appropriate cause.

2.3.1.8 Call Type Discrimination package(ctyp)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Properties	calltyp	O	
	ttyp	O	
	V8bsup	O	
	probemsg	O	used for text
	probeorder	O	used for text
Events	dtone	M	
Signals	v8sig	O	
	ans	O	
	callsig	O	
	v8bs	O	
	V18prob	O	Used for text

2.3.1.9 Base Root Package(root)

Attribute	Subfield	Mandatory/ Optional/ Not Required	Remarks
Properties	maxNumberOfContexts	O	
	maxTerminationsPerContext	O	
	normalMGCExecutionTime	O	

	provisionalResponseTimerValue	O	

2.3.1.10 RTP Package(rtp)

Attribute	**Subfield**	**Mandatory/ Optional/ Not Required**	**Remarks**
Statistics	ps	M	
	pr	M	
	pl	M	
	jit	M	
	delay	M	
Events	pltrans	O	

2.3.1.11 Inactivity Timer Package(it)

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

2.3.1.12 Overload Control Package(ocp)

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

2.4 Termination Naming Conventions

For digital trunks, it is proposed that the termination take the following format:-

ds/<unit-type1>-<unit #>/<unit-type2>-<unit #>/.../<channel #>

The <unit-type> identifies the particular hierarchy level (e.g. E1, STM1) and the <unit #> is a decimal number which is used to reference a particular instance of a <unit-type> at that level of the hierarchy. As stated in H248.1, the format of Termination identifiers is case- insensitive:

It is further recommended that MGs and Call Agents be flexible in terms of their naming conventions to facilitate inter-working.

2.5 Topology Descriptor

A gateway conforming to this profile is not required to implement Topology and MGCs expecting to control gateways meeting this specification shall not assume Topology is implemented.

2.6 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 MGs simultaneously. Support of virtual MGs as defined in H.248.1 Section 11.1 is optional.

2.7 Heartbeat

In order to detect loss of communication between MGC and MG, the Inactivity Timer package [8] shall be used. This mechanism is only used by the MG to detect loss of communication with MGC. Loss of communication can be due to:

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

On startup, the MG will initialize the timer to 0. The MGC shall use the Maximum Inactivity Time event to arm the MG. A value of 30 seconds is recommended for this timer. A value of 0 will disable the inactivity timer. Each time, the MG receives any message; it will restart the Inactivity timer. If the MGC has no messages to send, in order to prevent the inactivity timer expiring, the MGC may send an Audit on the root termination. This will be the only keep alive mechanism used. On the inactivity timer expiring, on the MG, the following actions shall be performed:

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

2.8 Startup Sequences

In order to avoid excessive messaging on startup, the following mechanism shall be supported:

- On MG restart, it shall send ServiceChange=Root, Method=Restart.
- On detecting MG restart, the MGC will send Reply ServiceChange=Root
- The MGC shall audit the status of all Terminations. To determine the status of physical terminations, the MGC shall send AuditValue on any terminationid. of each physical termination. For example to determine the status of all E1s the MGC shall send AuditValue=DS/S_01/STM1_01/E1_01/1 for each E1.
- The MG shall respond with Reply, Auditvalue=<terminationid>, ServiceState=InService or OutOfService. based on the ServiceState, the MGC shall build the status of all physical terminations.

2.9 Null Transaction Id

All TransactionReply 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 [1].

2.10 Transaction Timers

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

2.11 Transport

Gateways shall implement UDP and optionally implement SCTP[1] transport of H.248.

MGCs SHALL implement UDP and optionally SCTP transport of H.248.

Gateways and MGCs conforming to this profile are expected to transport H.248 signaling over IP.

2.12 Security

This release of the IA does not utilize security.

2.13 Encoding

Conforming Gateways SHALL support text encoding.

2.14 Timestamp

A timestamp SHALL be sent on every *Notify* message.

3 Voice Codecs

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

- G.711 (A-law or mu-law)

Support for additional codecs (e.g. G.723, G.729E) is optional.

The Trunking 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.

Silence suppression is optional although recommended unless it forms part of the codec (e.g. G.729B).

For the low bit rate codecs, e.g G.729, which are not capable of transmitting DTMF digits, telephony tones and signals in-band, digits, tones and signals shall be carried as RTP packets as specified in RFC 2833.

The Call Agent shall specify the codecs requested for the call in the Local and remote descriptors. The SDP parameter for codecs shall be specified using the Payload Type encodings specified in Table 4 of [4]. All codecs shall be identifed either by their fixed payload type (if applicable) or else their registered codec name (if using a dynamic type). See also [5].

4 Echo Cancellation

Trunking 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

[1] IETF RFC2960

all occurences. Echo Cancellation shall be applied to all voice calls, but turned off for voiceband data calls using G.711 upon detection of a 2100 Hz tone with phase reversal as per section 4.1 of G.168.

The Call Agent is responsible for ensuring far end echo cancellation occurs. Echo Cancellation is best performed as close to the echo source as possible and in many cases will be supported by an International Gateway, IntraLATA tandem switch or far-end User Agent. The Call Agent determines where the echo cancellation can be performed, for example via negotiation using SS7 signaling.

5 Modem, Fax and TTY Support

Trunking 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. Trunking 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.

Upon detection of the voiceband data, Trunking Gateways shall immediately switch to using the G.711 voice encoding until the end of the call. The codec switchover will be done by changing the RTP payload type for the RTP packets. When setting up the connection the Trunking Gateways and Call Agent shall ensure that G.711 is a supported codec in the session descriptor (SDP). In addition if there is an Notify command for fax or modem tones, the Trunking Gateways shall send a notification request.

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

As an option, Trunking Gateways and Call Agents may support T.38 fax transfer or V.152 modem transfer as an alternative to fallback to G.711.

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

Note, the latency in sending a Notification command for a fax/modem tone to the Call Agent followed by the Call Agent instructing the Trunking Gateways to switch codec is too long for modem calls to equalize at the maximum speed, hence the requirement for the Trunking Gateways to autonomously switch codecs as soon as it detects a fax or modem tone.

6 DTMF Digits and Telephony Tones

DTMF in-band digit support is out of scope for the GMI and thus section is optional.

If supported, DTMF digits shall be handled as specified in section 9 of [5].

In summary this specifies

- Media Gateways shall have the capability to transport DTMF digits using [6] 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 [6] or in-band. Support for [6] is specified using the RTP payload format ‘telephone-event’. If this payload format is specified in the SDP descriptor then [6] should be used if supported, otherwise in-band transport is used.

7 NAT and Firewall traversal

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

8 Redundancy and resilience

The Call Agent must support redundancy. This is required to match the existing PSTN five-nine's level of reliability. There are at least two ways to implement a redundant Call Agent with regards to the MEGACO signaling

- Implement a primary and secondary Call Agent as separate components in the network
  - Primary and backup Call Agents have different domain names
- Implement a Call Agent with no single point of failure and full redundancy

To support the first scheme, Trunking Gateways must support fail over to a secondary Call Agent if it detects a failure of the primary Call Agent, using the procedures described in section 11 of [1]. This procedure involves the Trunking Gateways having two or more Call Agents configured. In the event of losing contact with the primary Call Agent, the Trunking Gateways attempts to connect to the backup Call Agent(s) typically using a ServiceChange command. Synchronization and replication of data between the Primary Call Agent and Secondary Call Agent is outside the scope of this IA.

In the second scheme, the Call Agent has its own redundant components that share the same Domain name or IP address and the Trunking Gateways will be unaware that a failover has occurred.

In the case of the Trunking Gateway itself, the resilience requirements are dependent on the size of the card and thus the number of T1/E1s affected by the loss of such a card. Thus, if a card supports a few T1/E1s, then the individual T1/E1s can be distributed across different Trunk Groups such that loss of the card causes a small loss of capacity across a number of different interfaces. If the card supports many T1/E1s – e.g. an OC-12/STM-1 – then there must be some resilience built into the system (1+1 or 1:N redundancy of cards) since the loss of such a link would be unacceptable.

8.1 Timing Considerations

User Agents, Call Agents and Trunking Gateways need to derive their timing from the network. This is particularly vital to ensure that fax and modem traffic can be transported across the VoIP access network.

In the case of a Trunking Gateway, timing is typically received from an external source running over a physical interface. In addition, Trunking Gateways may also support an internal clock source (Stratum 3 or higher).

9 Management Information Model

9.1 Trunking Gateway

The following items must be configurable on the Trunking Gateway:

- User Agent Fully Qualified Domain Name or IP address
- DHCP or fixed IP address – typically Trunking Gateways would have a fixed IP address.
- If DHCP is not used
  - IP address
  - Subnet mask
  - Default IP gateway
- Name of Primary Call Agent
  - UDP Port to send MEGACO signaling to Primary Call Agent (default 2944)
- Name of Secondary Call Agent (optional)
  - UDP Port to send MEGACO signaling to Secondary Call Agent (default 2944)
- Heartbeat Timer value

- UDP Port to receive MEGACO signaling from Call Agent (default 2944)
- DSCP (TOS) byte value for MEGACO signaling traffic
- DSCP (TOS) byte value for RTP voice traffic

9.2 Call Agent

The following items must be configurable on the Call Agent

- Fully Qualified Domain Name
- DHCP or fixed IP address
- If DHCP is not used
    - IP address
    - Subnet mask
    - Default IP gateway
- UDP Port to receive MEGACO signaling from User Agents (default 2944)
- DSCP (TOS) byte value for MEGACO signaling traffic
- DSCP (TOS) byte value for RTP voice traffic
    - For each supported Trunking Gateway
    - Trunking Gateway Domain Name or IP address

UDP Port to send MEGACO signaling to User Agent (default 2944)

10 References

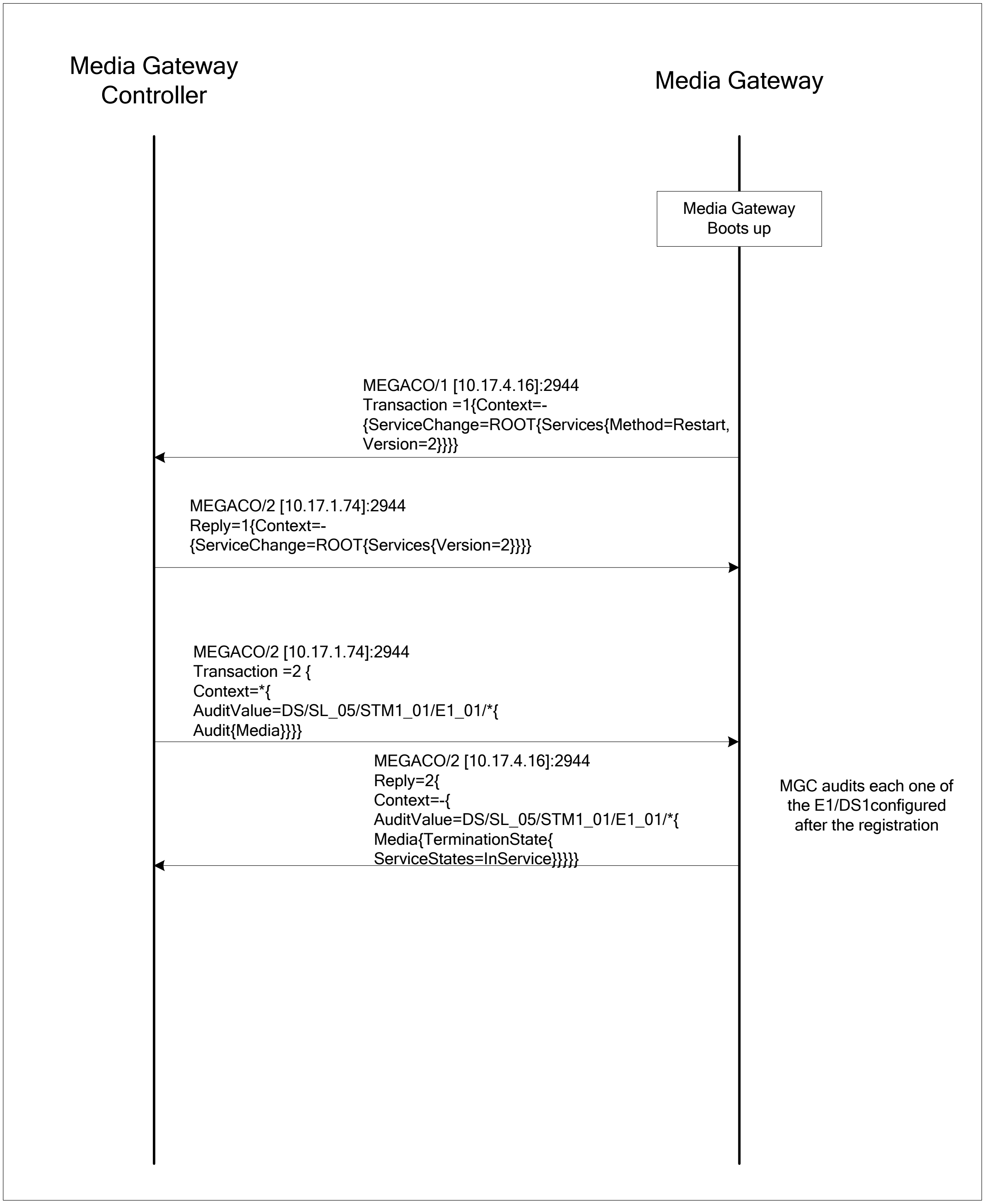
[1]	ITU-T H248.1: gateway Control Protocol Version 2
[2]	ITU-T H248.2: Gateway control protocol: Facsimile, text conversation and call discrimination packages
[3]	IETF – RFC 2327 SDP: Session Description Protocol, April 1998
[4]	IETF – RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control, July 2003
[5]	MSF-SDP-001-FINAL – IA for SDP Usage & Codec Negotiation
[6]	IETF – RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, May 2000
[7]	ITU-T H248.11: Media Gateway Overload Control Package
[8]	ITU-T H248.14: Inactivity Timer Package
[9]	ITU-T H248.8: Error Code & Service Change Reason Description.
[10]	ITU-T H248.13: Quality Alert Ceasing Package

Appendix A MEGACO Call Flows

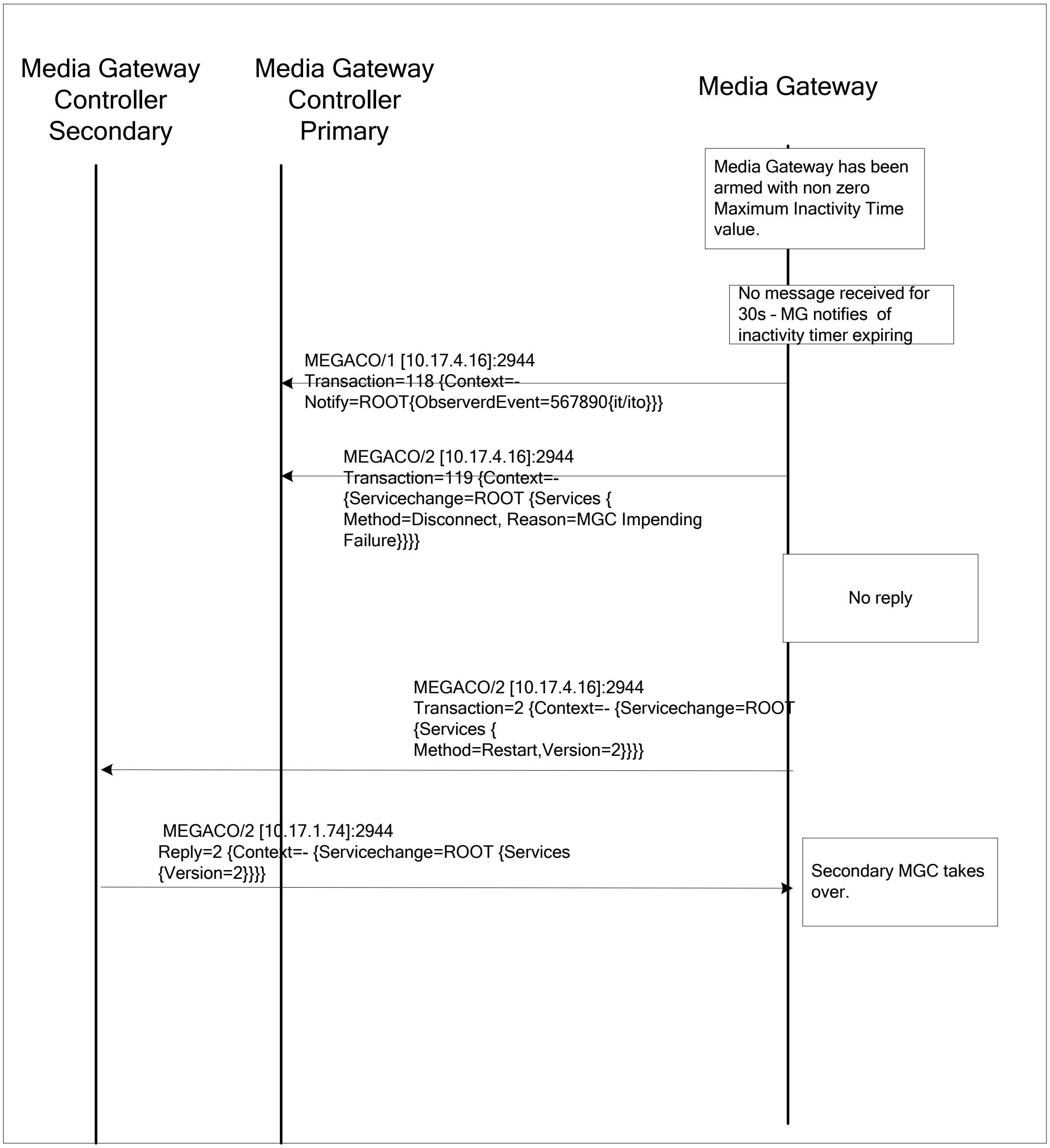
This appendix contains some example MEGACO call flows for various scenarios.

- MG Registration (Service Change Root)
- Service Change ROOT – Fail over
- Ingress Call
- Egress Call
- Call Disconnect
- COT Incoming call
- Modem Call
- Fax Call
- Inactivity Timer

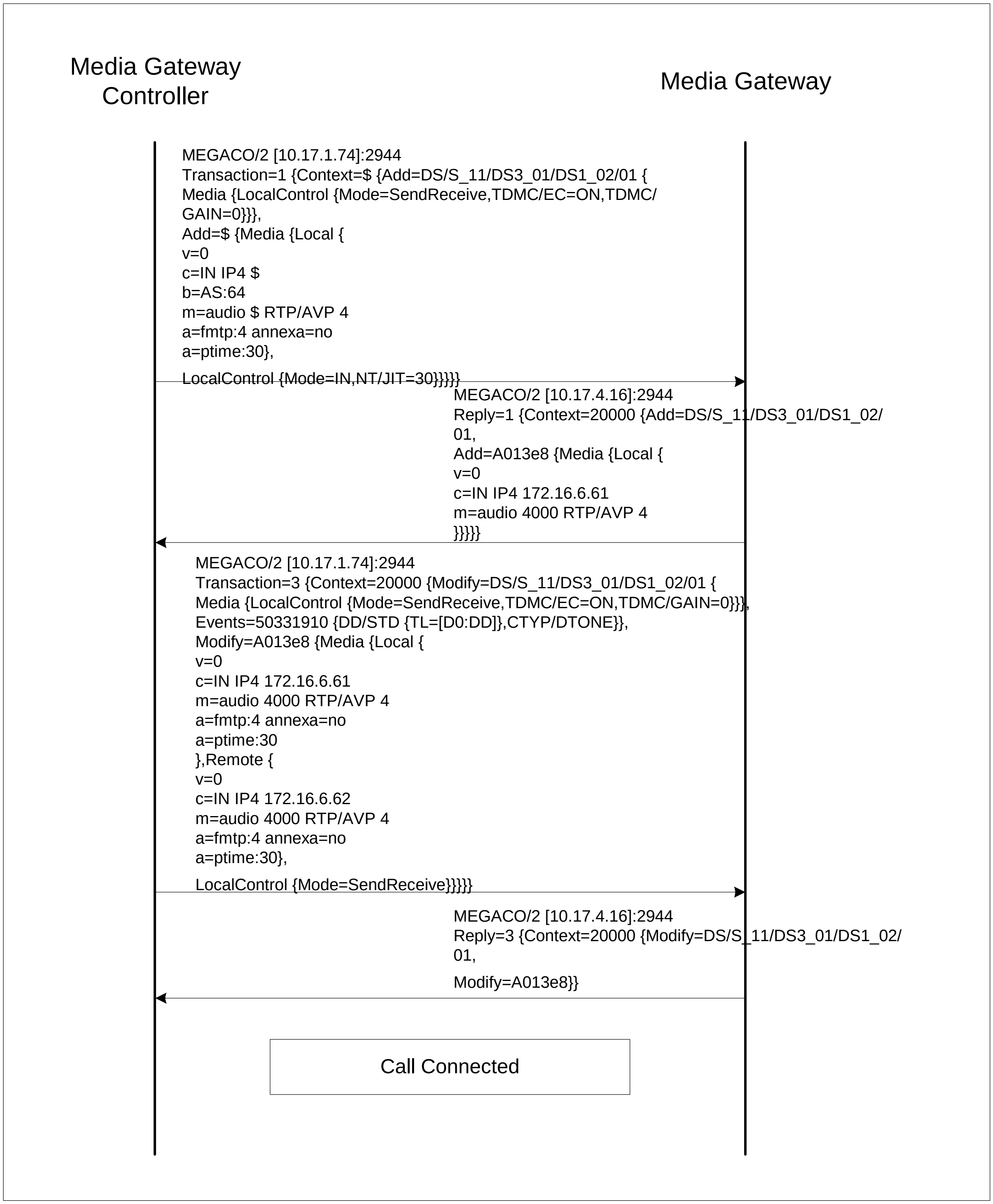
A.1 MG Registration (Service Change Root)



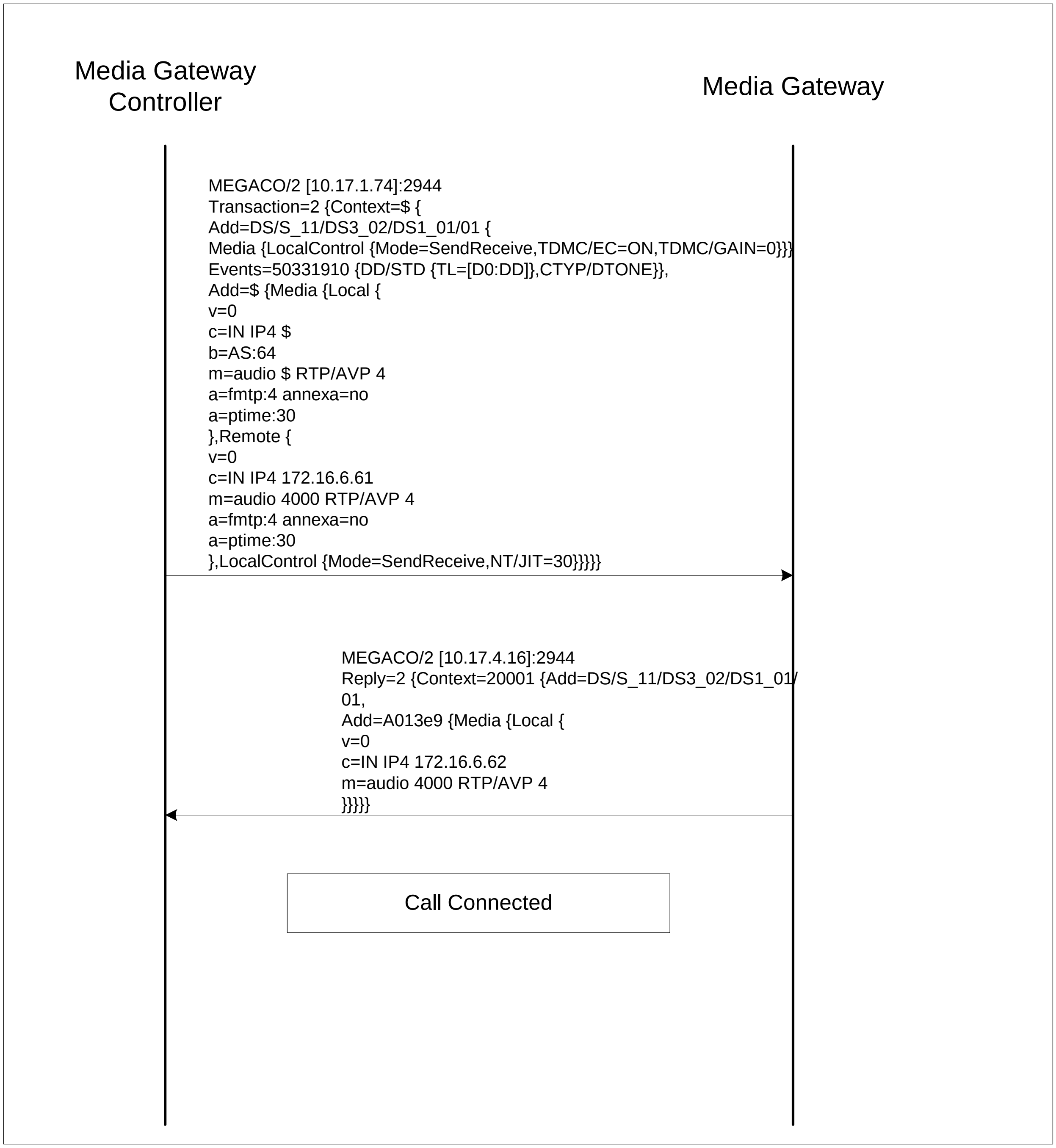
A.2 Service Change ROOT – Fail over



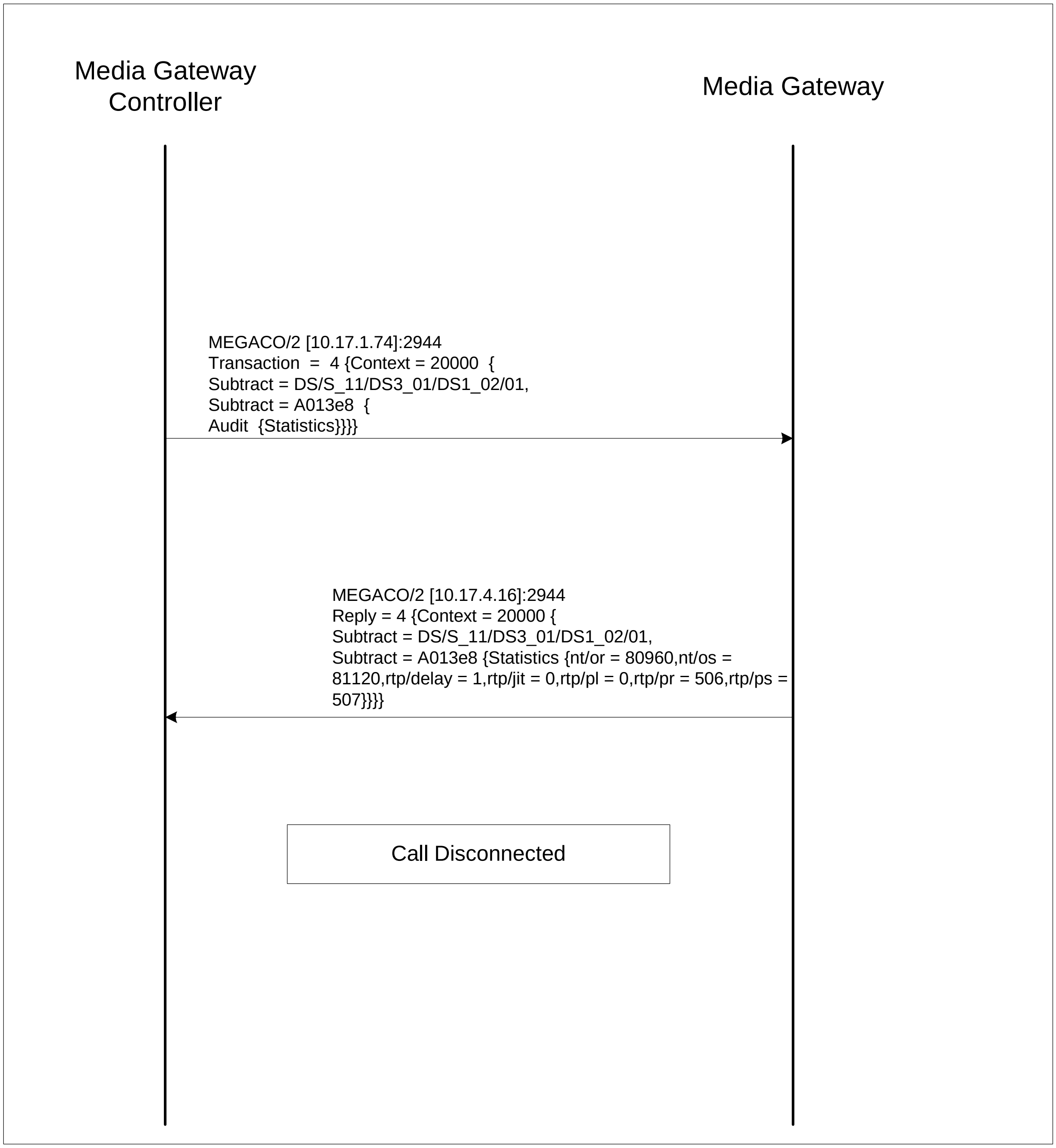
A.3 Ingress Call



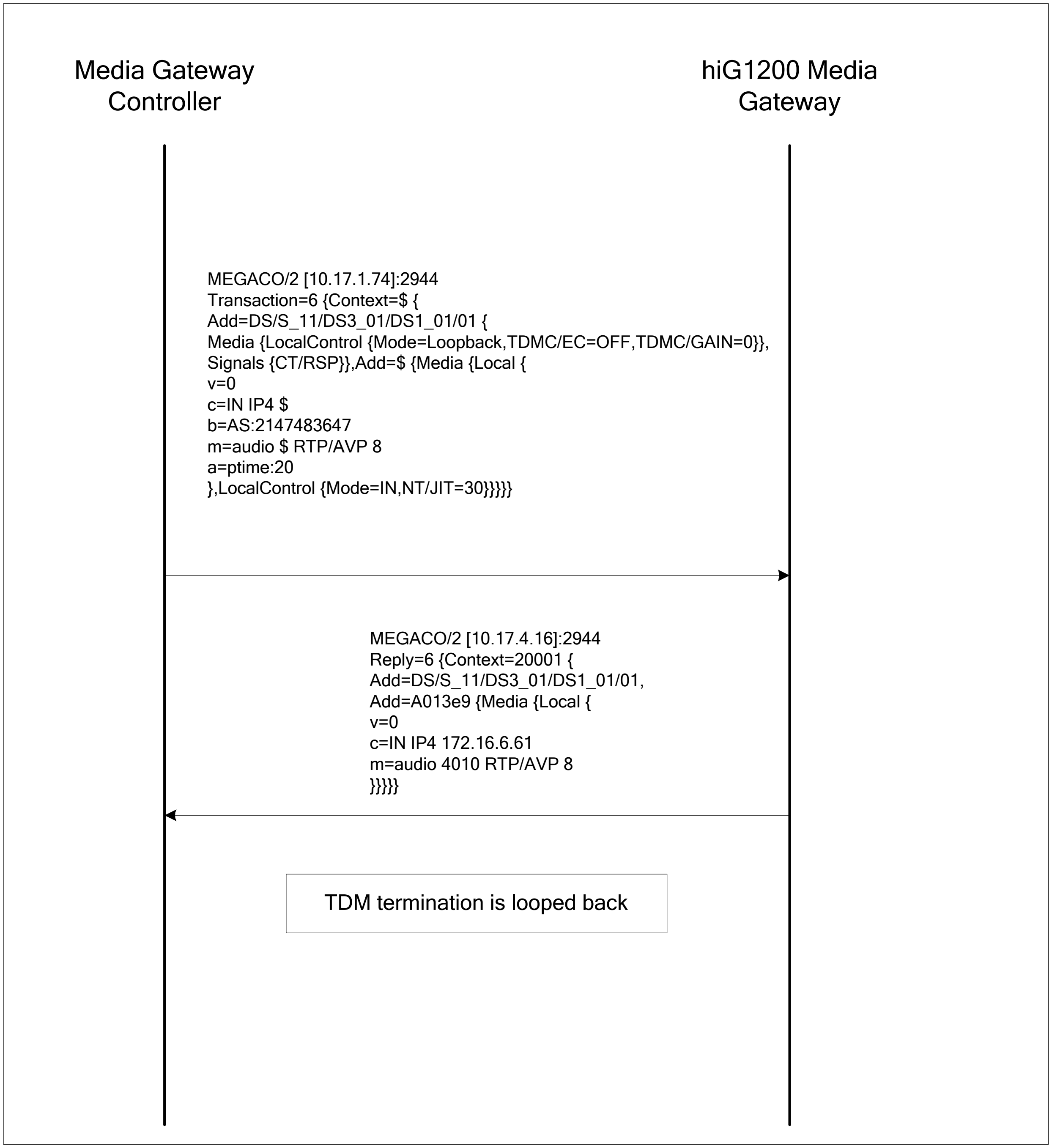
A.4 Egress Call



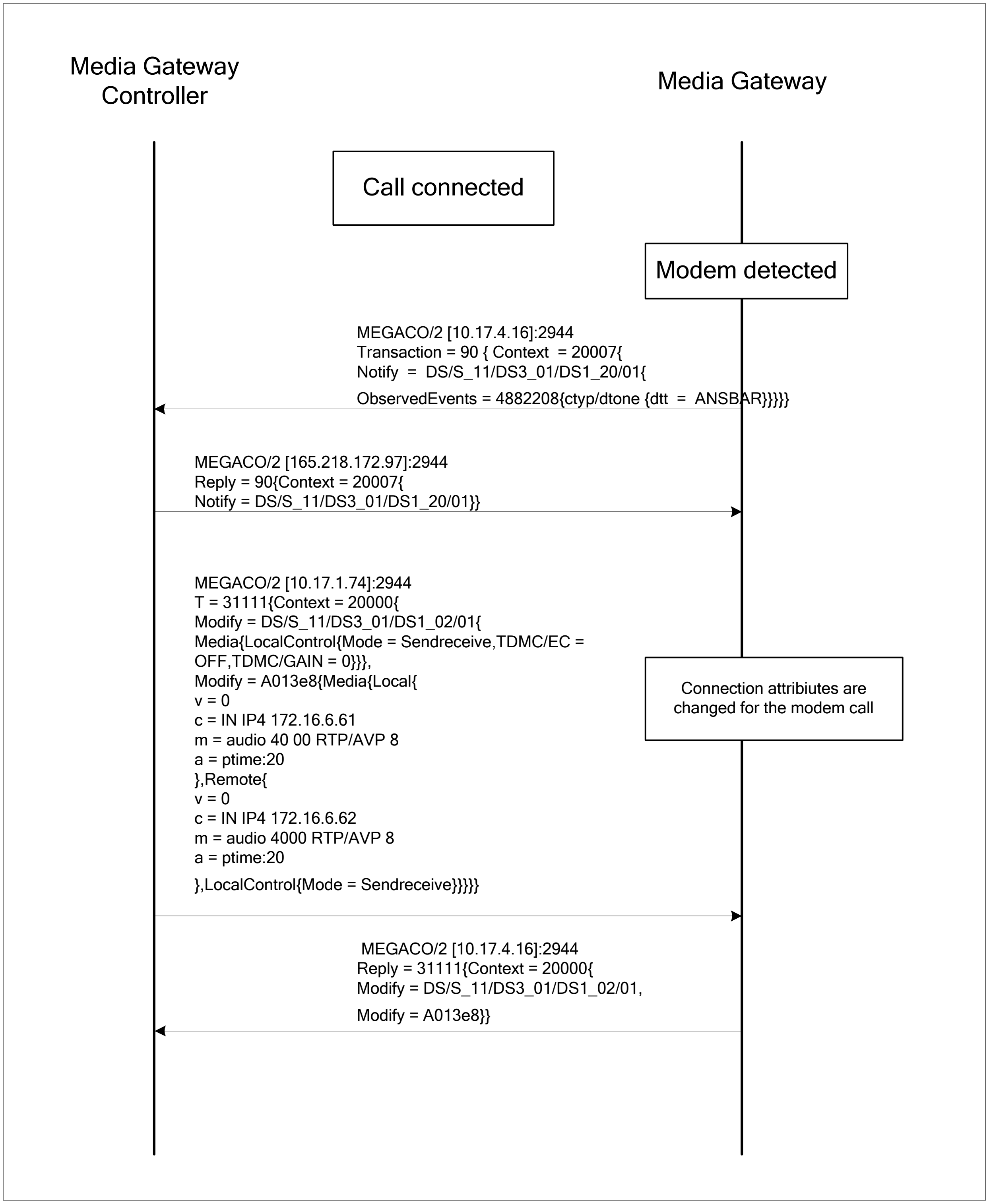
A.5 Call Disconnect



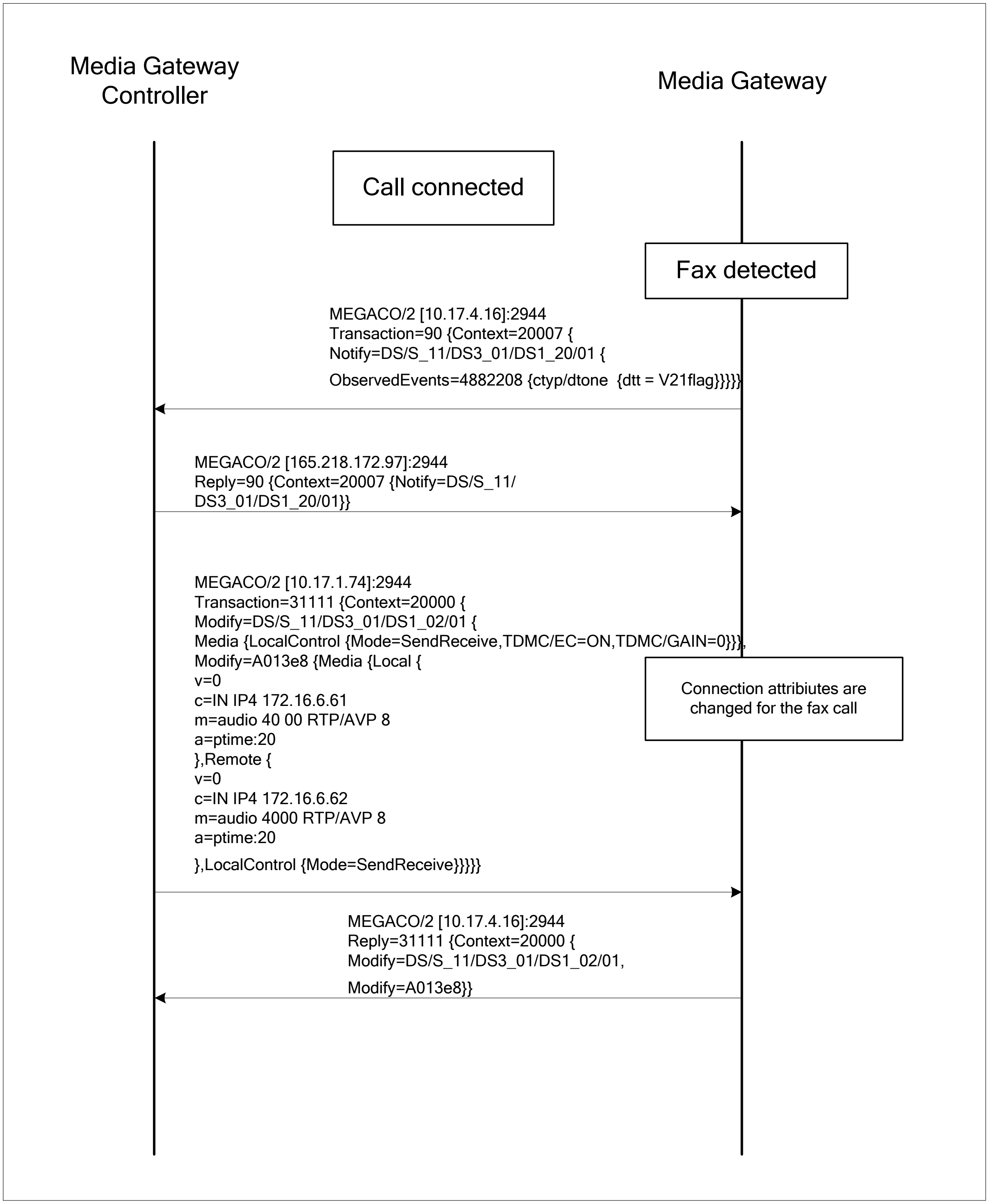
A.6 COT Incoming call



A.7 Modem Call



A.8 Fax Call


Media Gateway Controller
Media Gateway
Call connected
Fax detected
MEGACO/2 [10.17.4.16]:2944
Transaction=90 {Context=20007 {
Notify=DS/S_11/DS3_01/DS1_20/01 {
ObservedEvents=4882208 {ctyp/dtone {dtt = V21flag}}}}}
MEGACO/2 [165.218.172.97]:2944
Reply=90 {Context=20007 {Notify=DS/S_11/
DS3_01/DS1_20/01}}
MEGACO/2 [10.17.1.74]:2944
Transaction=31111 {Context=20000 {
Modify=DS/S_11/DS3_01/DS1_02/01 {
Media {LocalControl {Mode=SendReceive,TDMC/EC=ON,TDMC/GAIN=0}}},
Modify=A013e8 {Media {Local {
v=0
c=IN IP4 172.16.6.61
m=audio 40 00 RTP/AVP 8
a=ptime:20
},Remote {
v=0
c=IN IP4 172.16.6.62
m=audio 4000 RTP/AVP 8
a=ptime:20
},LocalControl {Mode=SendReceive}}}}}
Connection attribiutes are changed for the fax call
MEGACO/2 [10.17.4.16]:2944
Reply=31111 {Context=20000 {
Modify=DS/S_11/DS3_01/DS1_02/01,
Modify=A013e8}}

A.9 Inactivity Timer