ATIS-1000023.2013

ETS NETWORK ELEMENT REQUIREMENTS FOR NGN IMS-BASED DEPLOYMENTS

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ATIS-1000023.2013, *ETS Network Element Requirements for NGN IMS-based Deployments*

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American National Standard for Telecommunications

ETS Network Element Requirements for NGN IMS-based Deployments

Alliance for Telecommunications Industry Solutions

Approved August 12, 2013

American National Standards Institute, Inc.

Abstract

This document defines network element requirements to ensure that Emergency Telecommunications Service (ETS) is implementable and interoperable in a multi-vendor environment for an NGN IMS-based network deployment. These requirements further refine the procedures defined in the ETS in IP Networks Phase 1 standard [ATIS-1000010]. In addition, OA&M requirements are specified.

Foreword

The information contained in this Foreword is not part of this American National Standard (ANS) and has not been processed in accordance with ANSI's requirements for an ANS. As such, this Foreword may contain material that has not been subjected to public review or a consensus process. In addition, it does not contain requirements necessary for conformance to the Standard.

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ANSI guidelines specify two categories of requirements: mandatory and recommendation. The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following roster:

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The Signalling, Architecture, and Control (SAC) Subcommittee was responsible for the development of this document.

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1 Introduction

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2 Acronyms

A-BGF	Access Border Gateway Function
ACM	Address Complete Message
AIB	Authenticated Identity Body
AMR	Adaptive Multi-Rate
AMR-WB	Adaptive Multi-Rate Wideband
ANSI	American National Standards Institute
AS	Application Server
B2BUA	Back-to-Back User Agent
BE	Border Element
BFE	Border Functional Entity
BGCF	Border Gateway Control Function
CCFE	Call Control Functional Entity
CDMA	Code-Division Multiple Access
CdPN	Called Party Number
CPC	Calling Party's Category
CPG	Call Progress
codec	encoder/decoder
COS	Class of Service
DF	Default value
DHS/NCS	Department of Homeland Security/National Communications Service
DN	Destination Number
DSCP	Differentiated Services (DiffServ) Code Point
DSL	Digital Subscriber Line
DTMF	Dual-Tone Multi-Frequency
EF	Expedited Forwarding
EIP-GW	Egress IP Gateway
ETS	Emergency Telecommunications Service
ets.x	RPH name space for ETS with priority value x

FC	Feature Code
GETS	Government Emergency Telecommunications Service
GSM	Global System for Mobile Communications
GW	Gateway
HSS	Home Subscriber Service
IAM	Initial Address Message
IBCF	Interconnect Border Control Function
I-BGF	Interconnect Border Gateway Function
I-CSCF	Interrogating Call Session Control Function
IETF	Internet Engineering Task Force
IIP-GW	Ingress IP Gateway
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPBE	IP Border Element
ISDN	Integrated-Services Digital Network
ISTP	Internet Signaling Transport Protocol
ISUP	ISDN User Part
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
IWF	Interworking Function
MGCF	Media Gateway Control Function
MIME	Multipurpose Internet Mail Extensions
MLPP	Multi-Level Precedence and Preemption
MRB	Media Resource Broker
MRFC	Multimedia Resource Function Controller
MRFP	Multimedia Resource Function Processor
MS	Media Server
MS	Mobile Station
MSC	Mobile Switching Center
MTP	Message Transfer Part
NCS	National Communications System
NE	Network Element
NGN	Next Generation Network
NM	Network Management
NNI	Network to Network Interface
NS/EP	National Security/Emergency Preparedness
OA&M	Operations, Administration and Maintenance
P-CSCF	Proxy Call Session Control Function
PDF	Policy Decision Function
PHB	Per Hop Behavior
PIN	Personal Identification Number
PSTN	Public Switched Telephone Network
PSTNGW	PSTN Gateway
QSIG	Q signaling
RFC	Request for Comments

RP	Resource Priority
RPH	Resource Priority Header
RTP	Real-time Transport Protocol
SB	Service Broker
SBC	Session Border Controller
SCIM	Service Capability Interaction Manager
S-CSCF	Serving Call Session Control Function
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SGF	Signaling Gateway Function
S/I-CSCF	Serving/Interrogating Call Session Control Function
SIP	Session Initiation Protocol
SLF	Subscriber Locator Function
SS7	Signaling System No. 7
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TIA	Telecommunications Industry Association
T-MGF	Trunking Media Gateway Function
TR	Technical Report
VoIP	Voice over IP
UDP	User Datagram Protocol
UE	User Equipment
UMTS	Universal Mobile Telecommunications Service
UNI	User to Network Interface
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
WPS	Wireless Priority Service
wps.y	RPH name space for WPS with priority value y
WPS-FC	Wireless Priority Service Feature Code

3 Normative References

The following standards contain provisions which, through reference in this text, constitute provisions of this American National Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this American National Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

3.1 ATIS References

- ATIS-1000010.2006 (R2011), *Support of Emergency Telecommunications Service ETS in IP Network.*[1]
- ATIS-1000018, *NGN Architecture.*[2]

[1] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/product.aspx?id=25485 >

[2] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/product.aspx?id=22964 >

- ATIS-1000679.2013, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part.*[3]
- ATIS-1000005, *Service Description of ETS.*[4]
- ATIS-1000006.2005 (R201), *Signalling System No. 7 (SS7) - Emergency Telecommunications Service (ETS).*[5]
- ATIS-1000009.2006 (R2011), *IP Network-to-network (NNI) Standard for VoIP.*[6]

3.2 ITU[7]

- ITU-T Recommendation X.805, (10/03), *Security architecture for systems providing end-to-end communications*.
- ITU –T Recommendation Q.850, (05/98), *Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part.*

3.3 IETF References[8]

3.3.1 Call Control Signaling

- RFC 3261, *SIP: Session Initiation Protocol*, June 2002.
- RFC 3312, *Integration of Resource Management and Session Initiation Protocol (SIP)*, October 2002.
- RFC 3326, The *Reason Header Field for the Session Initiation Protocol (SIP)*, December 2002.
- < http://www.ietf.org/internet-drafts/draft-ietf-sip-resource-priority-08.txt > RFC 4412, *Communications Resource Priority for the Session Initiation Protocol (SIP)*, February 2006.
- RFC 4474, *Enhancements for Authenticated Identity Management in the Session Initiation Protocol (SIP)*, August 2006.
- RFC 4542, *Implementing an Emergency Telecommunications Service (ETS) for Real-Time Services in the Internet Protocol Suite*, May 2006.
- IETF RFC 2046 (1996), *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types.*
- IETF RFC 2327 (1998), *SDP: Session Description Protocol.*
- IETF RFC 2806 (2000), *URLs for Telephone Calls.*
- IETF RFC 2976 (2000), *The SIP INFO Method.*
- IETF RFC 3204 (2001), *MIME media types for ISUP and QSIG Objects.*
- IETF RFC 3264 (2002), *An Offer/Answer Model with the Session Description Protocol (SDP).*
- IETF RFC 3325 (2002), *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks.*
- IETF RFC 3323, *A Privacy Mechanism for the Session Initiation Protocol (SIP)*, November 2002.
- IETF RFC 3324, *Short Term Requirements for Network Asserted Identity*, November 2002.
- IETF RFC 3398, *Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping*, December 2002.
- IETF RFC 3420, *Internet Media Type message/sipfrag*, November 2002.

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[3] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/product.aspx?id=25371 >

[4] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/product.aspx?id=22553 >

[5] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/product.aspx?id=24944 >

[6] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/product.aspx?id=25486 >

[7] This document is available from the International Telecommunications Union. < http://www.itu.int/ITU-T/ >

[8] This document is available from the Internet Engineering Task Force (IETF). < http://www.ietf.org >

- IETF RFC 3428, *Session Initiation Protocol (SIP) Extension for Instant Messaging*, December 2002.
- IETF RFC 3515, *The Session Initiation Protocol (SIP) Refer Method*, April 2003.
- IETF RFC 3824, *Using E.164 numbers with the Session Initiation Protocol (SIP)*, June, 2004.
- IETF RFC 3966, *The tel URI for Telephone Calls*, December 2004.
- < draft-ietf-iptel-trunk-group-06.txt >, *Representing Trunk Groups in tel/sip URIs*, December 2005 (status: stable- AD watching).
- IETF RFC 3959, *The Early Session Disposition Type for the SIP*, December 2005.
- IETF RFC 3893, *SIP Authenticated Identity Body (AIB) Format*, September 2004. .
- IETF RFC 3911, *The Session Initiation Protocol (SIP) "Join" Header*, September 2004.
- IETF RFC 3892, *The SIP Referred-By Mechanism*, September 2004.
- IETF RFC 3891, *The Session Initiation Protocol (SIP) "Replaces" Header*, September 2004.
- IETF < http://www.ietf.org/internet-drafts/draft-ietf-sip-session-timer-14.txt > RFC 4028, *Session Timers in SIP*, February 2005.
- IETF RFC 3960, *Early Media and Ringback Tone Generation in the Session Initiation Protocol*, December 2004.
- IETF RFC 4694, *Number Portability Parameters for the "tel" URI*, October 2006.
- ETF RFC 3087, *Control of Service Context using SIP Request-URI*, April 2001.

3.3.2 Media references

- IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*.
- IETF RFC 3551 (2003), *RTP Profile for Audio and Video Conferences with Minimal Control*.
- IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.
- IETF RFC 3267 (2002), *Real-time Transport Protocol RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs*.
- IETF RFC 3389 (2002), *RTP Payload for Comfort Noise*.
- RFC 2475, *An Architecture for Differentiated Services*, December 1998.
- RFC 2597, *Assured Forwarding PHB Group*, June 1999.
- RFC 4733*, RTP Payload for DTMF Digits, Telephony Tones, and Telephony*. December 2006. (**Obsoletes RFC2833**)
- RFC 4734, *Definition of Events for Modem, Fax, and Text Telephony Signals*, December 2006. (**Obsoletes RFC2833**)
- RFC 3246, *An Expedited PHB (Per Hop Behavior)*, March 2002.
- RFC 3550 , *RTP: A Transport Protocol for Real-Time Applications*, July 2003.
- RFC 4594, *Configuration Guidelines for DiffServ Service Classes*, August 2006.
- IETF I-D (Work in Progress), draft-ietf-tsvwg-admitted-realtime-dscp-00, *An EF DSCP for Capacity-Admitted Traffic*, December 2006.
- IETF RFC 3551 (2003), *RTP Profile for Audio and Video Conferences with Minimal Control.*
- IETF RFC 3267 (2002), *Real-time Transport Protocol RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs.*
- IETF RFC 3389 (2002), *RTP Payload for Comfort Noise.*

3.4 Other

- 3GPP TR 22.952, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Priority service guide*; (Release 6).[9]

[9] This document is available from the Third Generation Partnership Project (3GPP) at
< http://www.3gpp.org/specs/specs.htm >.

- ANSI TIA-917, *Wireless Priority Service Enhancements for CDMA Systems*. [10]

4 Network Architecture Model

The ATIS NGN Architecture Technical Report [ATIS-1000018] defines a detailed functional architecture, based on the IMS architecture for the NGN, including definitions of the functional entities as shown in Figure 1. This architecture is independent of the services and access technologies being supported and can be instantiated in customized architectures designed in the context of specific services offered and the technologies used.

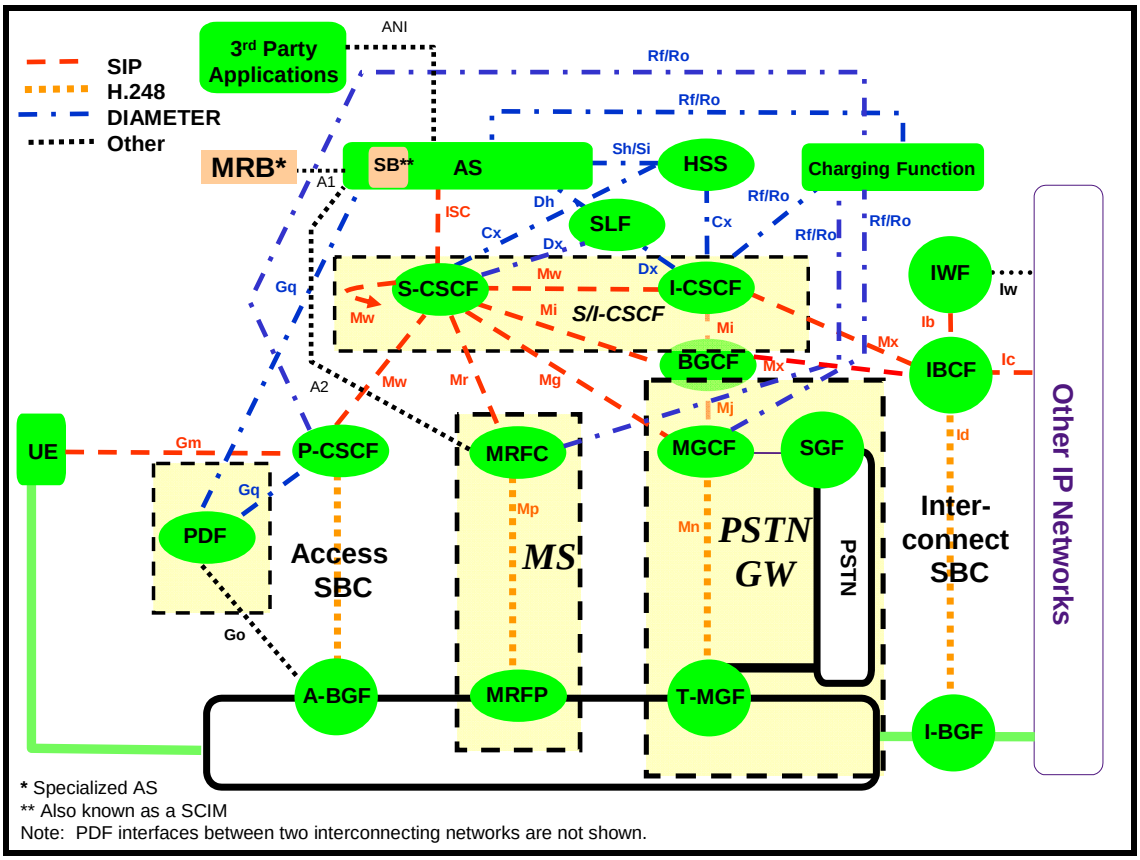

Figure 1– NGN Reference Architecture

[10] This document is available from the Telecommunications Industry Association (TIA).
< http://www.tiaonline.org/standards/overview.cfm >

The architecture as shown in Figure 1 bundles logical functions into network elements. The requirements in this document are for the following Network Elements:

Network Element	Functions
PSTN Gateway	MGCF, T-MGF, and SGF, and optional BGCF
IPBE (Access SBC)	P-CSCF & A-BGF , & PDF
IPBE (Interconnection SBC)	(IBCF)/ S/I-CSCF, I-BGF, & IWF
Media Server (MS)	MRFC & MRFP
AS	AS/SB & MRB
S/I-CSCF	S-CSCF & I-CSCF

5 Assumptions & General Principles

5.1 Assumptions

1. Within an IP domain, an ETS call/session is assigned one of five user priority levels associated with the calling service user.
2. SIP messages are only processed by SIP proxies/UAs, and therefore the SIP Resource Priority Header (RPH) is not processed by IP transport elements (e.g., routers).
3. Two types of core networks are considered in this Standard: IP-based and TDM-based. Within the TDM domain, ETS calls are called National Security/Emergency Preparedness (NS/EP) calls.
4. Three different originating and terminating access types are considered in this Standard: wireline IP-based (e.g., DSL, cable), wireline TDM-based, and wireless GSM/UMTS/CDMA-based.
5. An ETS call/session can traverse multiple IP and TDM core networks.
6. An ETS call/session originating over an IP access is authenticated either in an IP or a TDM core network. A WPS call is authenticated by the wireless service provider network of the calling party. IP-based wireless authentication (e.g., for UMTS) is not addressed in this document.
7. If both WPS and ETS authentication occur (i.e., the calling user dials WPS-FC+ETS-DN), the WPS authentication, within the wireless provider network, occurs before the ETS authentication.
8. The service provider's managed access and core networks are secure.
9. The DTMF tones are carried across an IP core network in accordance with RFC 4733, *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*.
10. The Network Elements implementing the Resource-Priority header in SIP signaling are configured to only recognize and act upon the ets and wps namespaces within the service provider's network. Other namespaces are passed in the Resource-Priority header but are not acted upon within the public network.
11. Use of other namespaces within private IP networks is not addressed. The processing of multiple namespaces, and the ordering of the namespaces among ets and other namespaces, as specified in IETF RFC 4412, Section 8, is for further study.
12. Use of feature codes (FCs) is for further study.
13. This document assumes UDP streams are used for signaling. Use of TCP and SCTP streams are for further study.

5.2 General Principles

1. Two types of authentication functions are considered in this Standard.

   - *ETS authentication* – This type of authentication is invoked when the calling user dials an ETS-DN.

     i. If the call is authenticated in the TDM domain, after the call is authenticated:

     - The CPC parameter in the IAM is set to "NS/EP Call".
     - Existing TDM authentication mechanisms do not have the calling user's priority value, so the Precedence parameter is not included in the IAM. If future authentication mechanisms have the calling user's priority value, the Precedence parameter will be included in the IAM.

     ii. If the call/session is authenticated in the IP domain, after the call/session is authenticated:

     - If the calling user's priority level is not available, only ets.x is created in the RPH, where x is based on policy[11]; or
     - If the calling user's priority level is available, both ets.x and wps.y namespaces are created in the RPH, where y is set to the calling user's priority level and x is based on policy.
     - The value for the ets namespace may be provisioned to be either a default value, or the value associated with the wps namespace.
     - The default of the provisionable value for the ets namespace is 0.
     - "DF" refers to the provisioned default value.

   - *WPS authentication* – This type of authentication is invoked when the calling user dials a WPS-FC+DN.

     i. If the call/session is authenticated in the TDM domain, after the call is authenticated:

     - The CPC parameter in the IAM is set to "NS/EP Call".
     - If the calling user's priority level is available, the Precedence parameter is included in the IAM and the Precedence level (in the Precedence parameter) indicates the user priority level (precedence level 0 – 4 corresponding to user priorities 1 – 5).

     WPS authentication in the IP domain (e.g., for UMTS phones) is not addressed in this document.

   Successful ETS authentication may result in changing the values in the ets or wps namespaces.

2. The ets namespace indicates an ETS (including a WPS) call/session.

   i. For a call traversing from a circuit-switched network to an IP network, the Ingress IP Gateway (IIP-GW) creates the SIP RPH ets namespace based on the presence of ETS-DN in the Called Party Number parameter or the Calling Party's Category parameter coded as "NS/EP Call" in the received IAM. The IIP-GW assigns the ets priority level. It is mandatory for the IIP-GW to support a provisionable default value for the ets priority level. The default value for the ets priority level is determined by policy. If no ISUP Precedence parameter is received, the default value is used to populate the ets priority level. If a Precedence parameter is received, then, based on policy, either the default value or the precedence level in the Precedence parameter is used to populate the ets priority level.

[11] Policy is defined by DHS/NCS.

- An ETS Authentication Application Server (AS) may modify the value of the received ets priority level based on a successful Authentication Verification Processing. An ETS Authentication Application Server (AS) rejects the call/session request if authentication or authorization is denied.
- If the call cannot be authenticated (e.g., due to an inability to access the authorization database in a timely manner), the Authentication AS treats the call/session request as authorized, and logs the request for subsequent analysis.
- If an ETS-compliant entity cannot access the ETS Authentication AS for call/session authorization (e.g., due to a network failure), it treats the call/session request as authorized, and logs the request for subsequent analysis.

3. A successfully authenticated WPS call/session includes the RPH with the ets namespace. In addition to the ets.x, the RPH for a successfully authenticated WPS call/session contains the "wps.y" namespace with the priority level (y) associated with the calling WPS user. For a WPS call/session traversing from a circuit-switched network to an IP network, the IIP-GW maps the precedence level in the received Precedence parameter, containing an approved WPS MLPP Service Domain, to the priority value in the wps namespace. The precedence level is determined during authentication of the WPS user. The presence of the ets namespace indicates an ETS or a WPS call/session and triggers priority treatment. However, the priority value (x) in the "ets.x" may be used for priority treatment only in the IP domain and is not used for priority treatment on the wireless access. The wps namespace value (y) is used for priority treatment on the wireless access.
4. The wps.y namespace is transported through the IP domain to facilitate priority treatment on the wireless egress access.
    i. An ETS Authentication Application Server (AS) may modify a received wps priority level or, if a wps namespace is not received, create a wps namespace based on Authentication Verification Processing.
    ii. For a call/session traversing from an IP network to a circuit-switched network, the Egress IP Gateway (EIP-GW) shall use the priority level in the wps namespace, if present, to populate the precedence level and MLPP Service Domain of the Precedence parameter in the outgoing ISUP IAM.
    iii. For a call/session traversing from an IP network to a circuit-switched network, if the ets namespace is present but the wps.y namespace is not present, the EIP-GW shall not send the Precedence parameter in the outgoing ISUP IAM.
5. Processing of ETS calls/sessions, including the associated signaling and media, are provided priority treatment over non-ETS calls based on the presence of ets.x.
6. The ets.x RPH priority value may be used for priority treatment of ETS traffic at certain interfaces, such as IP access-to-core and IP network-to-network interfaces, where connection admission control may be applied.
7. The ets.x RPH priority value may be used by SIP proxies and Back-to-Back User Agents (B2BUAs) to make routing decisions, particularly on IP access.
8. Based on policy, the Session Border Control function (CCFE and BFE) facing a user may modify a received RPH ets priority level by replacing the received level with the provisioned default value.
    i. The SBC function may reject a call with an ets RPH if the interface is not provisioned to accept call requests with an ets RPH. ETS calls using an ETS destination number are permitted across such interfaces.
9. All packets associated with an ETS call/session shall receive priority handling:
    i. In network element queues;
    ii. For access to the IP backbone; and
    iii. Within the IP backbone.

    This priority handling is based on the presence of an ets namespace in an RPH.

10. A secure mechanism that validates the identity of the far end sending network is required in order to support priority handling of packets on an IP-NNI.
11. The interaction between NEs in the processing of errored ets namespace RPHs was based on identifying errors during testing and normal operation by terminating the session request. This approach was seen as allowing the problem NE to be identified and fixed before an NS/EP event occurred. Removing the ets and wps namespaces and using the default ets namespace value are also specified as configuration options for errored headers. Selection of the appropriate configuration option shall be via contractual arrangement between the NCS and the service provider.

6 Network Element Protocol Requirements

6.1 General Requirements

All SIP capable VoIP elements support the following SIP RPH general requirements. These requirements address the general handling of SIP with RPH headers.

<VoIP-GEN-SIP-00100>

The SIP RPH capability shall be configurable as either enabled or disabled. The default is disabled.

<End of VoIP-GEN-SIP-00100>

<VoIP-GEN-SIP-00200>

An enabled SIP RPH capable VoIP NE (RPH VoIP NE) shall receive and act upon the SIP RPH header as specified in the NE specific requirements for that NE.

<End of VoIP-GEN-SIP-00200>

<VoIP-GEN-SIP-00300>

If a VoIP NE does not understand or process (e.g., the SIP RPH capability is disabled) the RPH field in a request or response, the VoIP NE shall ignore that header field, continue processing the message, and propagate the RPH unchanged.

<End of VoIP-GEN-SIP-00300>

<VoIP-GEN-SIP-00400>

An enabled SIP RPH capable VoIP NE shall format the Resource Priority Header field as follows:

- **Resource-Priority: namespace1.value1, namespace2.value2, …**

or

- **Resource-Priority: namespace1.value1**
- **Resource-Priority: namespace2.value2**

or

- **Resource-Priority: namespace1.value1, namespace3.value3**
- **Resource-Priority: namespace2.value2, ...**
- **...**

<End of VoIP-GEN-SIP-00400>

<VoIP-GEN-SIP-00500>

A particular namespace shall *not* appear more than once in the same SIP message.

<End of VoIP-GEN-SIP-00500>

<VoIP-GEN-SIP-00600>

For the ETS service, an ets namespace shall be present, with or without a wps namespace. The presence of a wps namespace without an ets namespace shall be processed as an error.

<End of VoIP-GEN-SIP-00600>

<VoIP-GEN-SIP-00700>

An enabled SIP RPH capable VoIP NE shall be able to generate the Accept-Resource-Priority header. The Accept-Resource-Priority field shall have the form:

Accept-Resource-Priority: namspace1.value1, namespace1.value2, …

<End of VoIP-GEN-SIP-00700>

NOTE: The "Accept-Resource-Priority" response header field enumerates the resource values (r-values) a SIP user agent server is willing to process.

<VoIP-GEN-SIP-00800>

For recognized ETS calls, an enabled SIP RPH capable VoIP NE shall insert the RPH into the following SIP messages:

INVITE, ACK, BYE, CANCEL, INFO, PRACK, REFER, UPDATE.

<End of VoIP-GEN-SIP-00800>

<VoIP-GEN-SIP-00900>

For recognized ETS calls, an enabled SIP RPH capable VoIP NE shall insert the RPH in 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx responses it sends, with the exception of 100 ("trying") responses and a 403 ("forbidden") message as specified in <VoIP-GEN-SIP-03200>. For messages associated with error conditions (e.g., 400 error messages with a 417 reason code, 417 messages, and 420 messages), the RPH shall carry only the ets.x. The RPH shall be inserted even if the received response from an upstream network element may not include an RPH.

<End of VoIP-GEN-SIP-00900>

<VoIP-GEN-SIP-01000>

An enabled SIP RPH capable VoIP NE shall send the "Supported" header field with the "resource-priority" option tag in an initial SIP request with the ets namespace.

<End of VoIP-GEN-SIP-01000>

NOTE: Due to potential interoperability problems, it is not recommended to send the "Required" header field with the "resource-priority" option tag.

<VoIP-GEN-SIP-01100>

A VoIP NE shall be able to receive a 182 (Queued) response from a VoIP NE that is currently at capacity, but that has put the original request into a queue.

<End of VoIP-GEN-SIP-01100>

<VoIP-GEN-SIP-01200>

When an enabled SIP RPH capable VoIP NE receives a 417 (Unknown Resource Priority) response or a 420 (Bad Extension) response with an "Unsupported: resource-priority" to the initial SIP request with the "Require" header field (including the "resource-priority" option tag) and the ets namespace, the enabled SIP RPH capable VoIP NE shall:

1. **Log the request and response as an error condition;**
2. **Attempt to send the request to the next VoIP NE in the route list; and**
3. **If all attempts receive a 417 response, the enabled SIP RPH capable VoIP NE shall resend the request without the "Require" header field, but with a "Supported" header field with the "resource-priority" option tag.**

<End of VoIP-GEN-SIP-01200>

<VoIP-GEN-SIP-01300>

For an enabled SIP RPH capable VoIP NE, use of the "resource-priority" option tag with the "Require" header field shall be configurable (i.e., enabled or disabled), with a default setting of disabled.

<End of VoIP-GEN-SIP-01300>

NOTE: According to RFC 3261, if a 420 (Bad Extension) response is received, the sending NE may retry the request, omitting any extensions listed in the Unsupported header field in the response. Removal of the Resource-Priority header will cause the request to be treated as a normal call at all downstream NEs. Thus, it is desirable that service providers analyze 420 responses received by an NE and configure the "route list" at that NE so that these routes are the last to be attempted.

NOTE: Requirements <VoIP-GEN-SIP-01400> through <VoIP-GEN-SIP-01800> are defined in support of the NEs only recognizing and acting upon the ets and wps namespace.

<VoIP-GEN-SIP-01400>

When an enabled SIP RPH capable VoIP NE receives a request with an RPH with multiple instances of the ets namespace (e.g., Resource-Priority: ets.0, ets.2), it shall:

1. **Log the header as an error condition; and**
2. **Reject the request with a 400 (Bad Request) response with a 417 code in the Reason header field. The response shall use the ets namespace with the DF value.**

<End of VoIP-GEN-SIP-01400>

<VoIP-GEN-SIP-01500>

When an enabled SIP RPH capable VoIP NE receives a request with an RPH with multiple instances of the wps namespace (e.g., Resource-Priority: ets.0, wps.2, wps.4), it shall:

1. **Log the header as an error condition; and**
2. **Reject the request with a 400 (Bad Request) response with a 417 code in the Reason header field. The response shall use the ets namespace with the DF value.**

<End of VoIP-GEN-SIP-01500>

<VoIP-GEN-SIP-01600>

When an enabled SIP RPH capable VoIP NE receives a request with an RPH with an invalid ets resource value (e.g., Resource-Priority: ets.7), it shall:

1. **Log the header as an error condition; and**
2. **Reject the request with a 400 (Bad Request) response with a 417 code in the Reason header field. The response shall use the ets namespace with the DF value.**

<End of VoIP-GEN-SIP-01600>

<VoIP-GEN-SIP-01700>

When an enabled SIP RPH capable VoIP NE receives a request with an RPH with an invalid wps resource value (e.g., Resource-Priority: ets.0, wps.9), it shall:

1. **Log the header as an error condition; and**
2. **Reject the request with a 400 (Bad Request) response with a 417 code in the Reason header field. The response shall not the ets namespace with the DF value.**

<End of VoIP-GEN-SIP-01700>

<VoIP-GEN-SIP-01800>

When an enabled SIP RPH capable VoIP NE receives a request with an RPH with a wps resource value and no ets resource value (e.g., Resource-Priority: wps.4), it shall:

1. **Log the header as an error condition; and**
2. **Reject the request with a 400 (Bad Request) response with a 417 code in the Reason header field. The response shall use the ets namespace with the DF value.**

<End of VoIP-GEN-SIP-01800>

NOTE: An NE needs to identify only one error associated with the ets and wps namespaces, and not all errors. This is so the NE does not send back multiple 400 responses, one for each error discovered. The log will contain the complete header, which can be analyzed later.

<VoIP-GEN-SIP-01900>

When an enabled SIP RPH capable VoIP NE receives a 400 (Bad Request) response with a 417 code in the Reason header field, the enabled SIP RPH capable VoIP NE shall be configurable to:

1. **Reject the call; or**
2. **Resend the call without the ets and wps namespaces, preserving the other namespaces in the RPH; or**
3. **Resend the call with a single copy of ets.DF, removing the wps namespace if present, and preserving the other namespaces in the RPH.**

The default action is to resend the call with a single copy of ets.DF, removing the wps namespace if present, and preserving the other namespaces in the RPH.

<End of VoIP-GEN-SIP-01900>

<VoIP-GEN-SIP-02000>

Upon completion of one of the actions in Requirement VoIP-GEN-SIP-01900, the VoIP NE shall log the request and response as an error condition

<End of VoIP-GEN-SIP-02000>

<VoIP-GEN-SIP-02100>

An enabled SIP RPH capable VoIP NE receiving a SIP message with an RPH shall propagate the RPH in the message and shall insert the RPH in each response it generates to the received message. The exceptions to this requirement are the 100 and 403 responses, which do not have an RPH.

<End of VoIP-GEN-SIP-02100>

<VoIP-GEN-SIP-02200>

An enabled SIP RPH capable VoIP NE receiving a SIP response with an RPH shall propagate the RPH in the response and shall insert the RPH in new messages it generates associated with this session.

<End of VoIP-GEN-SIP-02200>

<VoIP-GEN-SIP-02300>

For recognized ETS calls/sessions, an enabled SIP RPH capable VoIP NE shall receive and provide priority treatment based on the RPH contained in the following SIP messages:

INVITE, ACK, BYE, CANCEL, INFO, PRACK, REFER, UPDATE.

<End of VoIP-GEN-SIP-02300>

<VoIP-GEN-SIP-02400>

An enabled SIP RPH capable VoIP NE shall also receive and provide priority treatment for the 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx responses for recognized ETS calls (sessions), except for 100 and 403 responses. This includes the case where the upstream network element may not respond with an RPH included.

<End of VoIP-GEN-SIP-02400>

<VoIP-GEN-SIP-02500>

When an enabled SIP RPH capable VoIP NE receives an ACK, BYE, CANCEL, INFO, PRACK, REFER, UPDATE or a response with an ets RPH for a non-recognized ETS call/session (i.e., the INVITE did not have an ets namespace or an ETS-DN), the enabled SIP RPH capable VoIP NE shall:

1. **Log the message as an error condition;**
2. **Use neither the ets nor wps namespaces in messages and responses associated with this session; and**
3. **Process the messages and responses as for a normal call.**

<End of VoIP-GEN-SIP-02500>

<VoIP-GEN-SIP-02600>

If an enabled SIP RPH capable VoIP NE does not understand any of the resource values in the request, the treatment depends on the presence of the "Require" "resource-priority" option tag:

1. **Without the option tag, the enabled SIP RPH capable VoIP NE shall treat the request as if it contained no "Resource-Priority" header field and processes the request as a normal call/session. Resource values that are not understood shall *not* be modified or deleted. Resource values that are understood to be in error shall be handled as specified in <VoIP-GEN-SIP-01400> thru <01800>.**
2. **With the option tag, it shall reject the request with a 417 (Unknown Resource-Priority) response code.**

A 417 (Unknown Resource-Priority) response *may*, if specified according to local policy, include an "Accept-Resource-Priority" header field enumerating the acceptable resource values.

Sending the "Accept-Resource-Priority" header field with an Unknown Resource-Priority (417) response shall be configurable (enabled or disabled), with a default value of disabled.

If a VoIP NE receives a SIP request containing the "Require" header field with the "resource-priority" option tag, there are three possible responses:

1. **Following standard SIP behavior, if a SIP request contains the "Require" header field with the "resource-priority" option tag, a SIP user agent shall respond with a 420 (Bad Extension) if it does not support the SIP RPH extension. It then lists "resource-priority" in the "Unsupported" header field included in the response.**
2. **If the NE is an enabled SIP RPH capable VoIP NE, but does not understand any of the namespaces, then it shall reject the request with a 417 (Unknown Resource-Priority) response code. Namespaces that are understood to be in error (e.g., incorrect syntax) shall be handled as specified in Requirements <VoIP-GEN-SIP-01400> thru <01800>.**
3. **If the NE is an enabled SIP RPH capable VoIP NE, understands the ets namespace, and the namespace syntax is correct, it shall process the SIP request in accordance with the requirements found in this document.**

<End of VoIP-GEN-SIP-02600>

<VoIP-GEN-SIP-02700>

If a SIP request fails because an enabled SIP RPH capable VoIP NE cannot handle the signaling load, the NE shall return a 503 (Service Unavailable) response. The response shall include a Reason Header Q.850 value of 42 (switching equipment congestion).

<End of VoIP-GEN-SIP-02700>

<VoIP-GEN-SIP-02800>

When a call request arrives at an enabled SIP RPH capable VoIP NE that is unable to accept the call (e.g., due to processor resource constraints), the UAS shall return a 503 (Service unavailable) response. This response shall include a Reason Header Q.850 value of 42 (switching equipment congestion).

<End of VoIP-GEN-SIP-02800>

<VoIP-GEN-SIP-02900>

If an enabled SIP RPH capable VoIP NE gets 486 (Busy Here) responses on all branches, as identified by the branch parameter in the Via Header, and has exhausted all available call treatments, it shall then return a 600 (Busy Everywhere) response to the caller. The response shall include a Reason Header Q.850 value of 17 (user busy).

<End of VoIP-GEN-SIP-02900>

<VoIP-GEN-SIP-03000>

An enabled SIP RPH capable VoIP NE shall not change the Unsupported header value for RP.

<End of VoIP-GEN-SIP-03000>

NOTE: If an NE receives a 503, it attempts the request to the next VoIP NE in the route list. Though this is correct SIP behavior, it needs to be validated in the architecture.

<VoIP-GEN-SIP-03100>

An enabled SIP RPH capable VoIP NE shall be configurable on a per-UNI and per-NNI basis to either:

1. **Use the ets resource value received across an IP UNI or an IP NNI in requests it sends into the network; or**
2. **Reset the received ets resource value to the DF resource value for requests sent into the network.**

<End of VoIP-GEN-SIP-03100>

<VoIP-GEN-SIP-03200>

Each IP UNI and NNI on a SIP RPH capable VoIP NE shall be configurable to accept or reject a SIP INVITE with the ets RPH and a normal (non-ETS-DN) destination number. The default shall be to reject the SIP INVITE. It is recommended that the reject be a 403 (Forbidden) response with two reason header fields: (1) a 417 (RPH header) code for SIP; and (2) a Reason Header Q.850 value of 21 (call rejected). Note that the 403 response shall not contain
the ets RPH header.

If the UNI or NNI is provisioned to accept an ets RPH, then <VoIP-GEN-SIP-03100> shall apply.

<End of VoIP-GEN-SIP-03200>

NOTE: <VoIP-GEN-SIP-03200> refers specifically to a SIP INVITE and not to other messages associated with a session. <VoIP-GEN-SIP-02500> handles the case for an ets Resource-Priority header in other messages.

6.2 PSTN Gateways (PSTNGW)

Assumptions:

1. The PSTNGW Border Elements (BEs) are connected to an IP Backbone Edge Router by a High Capacity Ethernet interface.
2. These PTSNGW BEs support both TDM SS7 ISUP and IP SIP.

<VoIP-PSTNGW-SIP-00100>

Upon receipt of an ISUP IAM with CdPN =ETS-DN and/or CPC=NSEP, for subsequent SIP messages in response to the IAM, the PSTNGW shall create the RPH with an ets namespace. The ets value shall have a default value; the default shall be configurable. Also, the PSTNGW shall copy the precedence level into the wps namespace if the MLPP service domain indicates a wps call.

<End of VoIP-PSTNGW-SIP-00100>

<VoIP-PSTNGW-SIP-00200>

The PSTNGW shall allow the ets.x value to be assigned the precedence level with proper MLPP service domain information. This action shall be configurable.

<End of VoIP-PSTNGW-SIP-00200>

<VoIP-PSTNGW-SIP-00300>

VoIP PSTNGWs that have TDM NS/EP (i.e., GETS or WPS) features triggered on the receipt of an SS7 ISUP IAM with CPC=NS/EP call or on a GETS DN shall trigger all such NS/EP features upon receipt of an ets.x RPH or on a SIP INVITE with an ETS-DN.

<End of VoIP-PSTNGW-SIP-00300>

<VoIP-PSTNGW-SIP-00400>

VoIP PSTNGWs that have implemented VoIP NS/EP priority treatment features shall trigger these features based upon the receipt of RPH=ets.x or a SIP INVITE with To header = ETS-DN.

<End of VoIP-PSTNGW-SIP-00400>

<VoIP-PSTNGW-SIP-00500>

VoIP PSTNGWs that have implemented VoIP NS/EP priority treatment features triggered on the receipt of an RPH=ets.x or a SIP INVITE with To header = ETS-DN shall trigger all such NS/EP features upon receipt of an SS7 ISUP IAM with CPC=NS/EP call or a GETS DN

<End of VoIP-PSTNGW-SIP-00500>

<VoIP-PSTNGW-SIP-00600>

An PSTNGW NE acting as a PSTN/IP gateway or as an IP/PSTN gateway shall recognize an incoming ETS call by:

a) **The presence of a valid RPH of ets.x; or**
b) **The presence of an ETS-DN (e.g., 710-NXX-XXXX); or**
c) **The presence of a CPC value of NS/EP call.**

<End of VoIP-PSTNGW-SIP-00600>

NOTE: The intent of this requirement is that any combination of these shall cause the call to be recognized as an ETS call.

<VoIP-PSTNGW-SIP-00700>

A minimum of 10 ETS-DNs shall be configurable or provisionable. A DN may be a partial string (e.g., 710) or a complete number (e.g., 710 NCS GETS). Matching of the ETS-DNs shall start with the leading digit of the received called number.

<End of VoIP-PSTNGW-SIP-00700>

NOTE: NS/EP feature code matching at the PSTNGW is for further study.

<VoIP-PSTNGW-SIP-00800>

The PSTNGW shall use the following table to determine the incoming ISUP ETS-DN, CPC and Precedence parameter value mappings to the outgoing SIP message RPH values for ets.x and wps.y

ISUP	SIP
CdPN = Destination Number CPC not equal to NS/EP Call No PRECEDENCE PARAMETER This is a normal (non-ETS) call	R-URI/To: = Destination Number
CdPN = ETS-DN CPC = NS/EP Call No PRECEDENCE PARAMETER	R-URI/To: = ETS-DN RPH [ets.DF]

ISUP	SIP
CdPN = Destination Number CPC = NS/EP Call No PRECEDENCE PARAMETER	R-URI/To: = Destination Number RPH [ets.DF]
CdPN = ETS-DN CPC not equal to NS/EP Call No PRECEDENCE PARAMETER	R-URI/To: = ETS-DN RPH [ets.DF]
CdPN = Destination Number CPC not equal to NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H‘40024B’ - H‘40024F’)	Request for ETS handling is rejected, and the call is processed as an ordinary call.
CdPN = Destination Number CPC = NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H‘40024B’ - H‘40024F’)	R-URI/To: = Destination Number RPH [(ets.DF, wps.y) or (ets.y, wps.y)]
CdPN = ETS-DN CPC = NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H‘40024B’ - H‘40024F’)	R-URI/To: = ETS-DN RPH [(ets.DF, wps.y) or (ets.y, wps.y)]
CdPN = ETS-DN CPC not equal to NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10	R-URI/To: = ETS-DN RPH [ets.DF, wps.y]

ISUP	SIP
Precedence Level = y (0 – 4) **Network Identity = 0100** **MLPP Service Domain = z (H'40024B' - H'40024F')** **This is abnormal and shall be logged.**	
The default (DF) priority level is provisioned at the PSTN Gateway and its value is determined by policy. The default priority level is used until the call can be authenticated. **The NS/EP Call value for CPC is H'E2'** **Note that if y = 0 then z = H'40024B', and if y = 4 then z = H'40024F'.**	

<End of VoIP-PSTNGW-SIP-00800>

<VoIP-PSTNGW-SIP-00900>

Upon receipt of a 182 response, a VoIP PSTNGW shall determine if an ISUP ACM message has already been sent for this ETS request:

a) **If an ISUP ACM message was not previously sent, an ISUP ACM message with a Called Party Status indicator in the Backward Call Indicators parameter set to binary value 11 ("excessive delay") shall be sent; or**
b) **If an ISUP ACM message was previously sent, an ISUP CPG message with a Called Party Status indicator in the Backward Call Indicators parameter set to binary value 11 ("excessive delay") shall be sent.**

<End of VoIP-PSTNGW-SIP-00900>

<VoIP-PSTNGW-SIP-01000>

This requirement applies to the UNI.

If PSTNGW does not have enough bandwidth on UE trunks/connections, or if there is an insufficient number of trunks, then a 488 (Not Acceptable Here) response shall be sent. The 488 response shall include a "Warning" header field with a reason for the rejection; warning code 370 (Insufficient Bandwidth). This response shall include a Reason Header Q.850 value of 34 (no circuits available) In addition:

a) **If the request is queued for media resources and the request exceeds the maximum configured waiting time in queue, then the PSTNGW shall return 408 (Request Timeout). This response shall include a Reason Header Q.850 value of 102 (recovery of timer expiry).**
b) **If the PSTNGW is unable to queue the call, the PSTNGW shall return a 486 (Busy here) response. The response shall include a Reason Header Q.850 value of 17 (user busy).**

<End of VoIP-PSTNGW-SIP-01000>

<VoIP-PSTNGW-SIP-01100>

This requirement applies to the NNI.

If PSTNGW does not have enough bandwidth on interswitch trunks/connections, or if there is an insufficient number of trunks, then a 503 (Service Unavailable) response shall be sent. This response shall include a Reason Header Q.850 value of 34 (no circuits available). In addition:

a) **If the request is queued for media resources and the request exceeds the maximum configured waiting time in queue, then the PSTNGW shall return 408 (Request Timeout) when the request exceeds the maximum configured waiting time in queue for the last route available. The Reason Header Q.850 value associated with this response shall be provisionable, with a default 102 (recovery of timer expiry).**
b) **If the PSTNGW is unable to queue the call, the PSTNGW shall return a 503 (Service Unavailable) response. The response shall include a Reason Header Q.850 value of 34 (no circuit available).**

<End of VoIP-PSTNGW-SIP-01100>

<VoIP-PSTNGW-SIP-01200>

If a PSTNGW's resources are busy and the request is queued, it shall generate a 182 (Queued) response, until it is able to handle the request and provide a final response. The frequency of such provisional messages is governed by [RFC 3261].

<End of VoIP-PSTNGW-SIP-01200>

<VoIP-PSTNGW-SIP-01300>

If PSTNGW cannot support a call request due to an incompatibility between the offered and available codecs, the PSTNGW shall generate a 488 response (Not Acceptable Here). The response shall include a Reason Header Q.850 value of 29 (Facility Rejected).

<End of VoIP-PSTNGW-SIP-01300>

<VoIP-PSTNGW-SIP-01400>

If PSTNGW cannot support a call request due to an inability to reach the requested IP address, the PSTNGW shall generate a 488 response (Not Acceptable Here). The response shall include a Reason Header Q.850 value of 3 (No Route).

<End of VoIP-PSTNGW-SIP-01400>

<VoIP-PSTNGW-SIP-01500>

For a call traversing from an IP network to a circuit-switched network, the Egress PSTNGW shall use the priority level in the wps namespace, if present, to populate the precedence level and MLPP Service Domain of the Precedence parameter in the outgoing ISUP IAM.

For a call traversing from an IP network to a circuit-switched network, if the ets.x namespace is present but the wps.x namespace is not present, the PSTNGW shall not send the Precedence parameter in the outgoing ISUP IAM.

<End of VoIP-PSTNGW-SIP-01500>

<VoIP-PSTNGW-SIP-01600>

A PSTNGW shall use the following table for mapping from SIP RPH = ets.x, wps.y to ISUP CPC and Precedence parameter.

SIP	ISUP
R-URI/To: = Destination Number This is a normal (non-ETS) call	CdPN = Destination Number CPC not equal to NS/EP Call MTP PRIORITY = 0 No PRECEDENCE PARAMETER
R-URI/To: = Destination Number RPH [ets.x] If from trusted source. See <VoIP-PSTNGW-SIP-01700> for non-trusted source	CdPN = Destination Number CPC = NS/EP Call, MTP PRIORITY = 1 No PRECEDENCE PARAMETER
R-URI/To: = Destination Number RPH [ets.x, wps.y] If from trusted source. See <VoIP-PSTNGW-SIP-01700> for non-trusted source	CdPN = Destination Number CPC = NS/EP Call, MTP PRIORITY = 1 PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H‘40024B’ - H‘40024F’)
R-URI/To: = ETS-DN No RPH or ets.DF	CdPN = ETS-DN CPC = NS/EP Call, MTP PRIORITY = 1 No PRECEDENCE PARAMETER
R-URI/To: = ETS-DN RPH [ets.x, wps.y]	CdPN = ETS-DN CPC = NS/EP Call, MTP PRIORITY = 1 PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H‘40024B’ - H‘40024F’)
R-URI/To: = ETS-DN RPH [wps.y] This is an errored header	Request is rejected using 400 response with 417. See <VoIP-GEN-SIP-01800>
R-URI/To: = Destination Number RPH [wps.y]	Request is rejected using 400 response with 417. See <VoIP-GEN-SIP-01800>

SIP	ISUP
This is an errored header	
The default (DF) priority level is provisioned at the PSTN Gateway and its value is determined by policy. The default priority level is used until the call can be authenticated. **The NS/EP Call value for CPC is H'E2'.** **Note that if y = 0 then z = H'40024B', and if y = 4 then z = H'40024F'.**	

<End of VoIP-PSTNGW-SIP-01600>

<VoIP-PSTNGW-SIP-01700>

A request received with an ETS-DN is assumed to be for an ETS (NS/EP) call, and the device shall provide NS/EP treatment to the call. The PSTNGW may set or modify the values associated with the ets and wps namespaces, based on provisionable parameters.

The PSTNGW shall be configurable to accept or reject a SIP INVITE with the ets RPH and a normal (non-ETS-DN) destination number on a per interface basis. The default shall be to reject the SIP INVITE. It is recommended that the reject be a 403 (Forbidden) response with two reason header fields: (1) a 417 (RPH header) code for SIP; and (2) a Reason Header Q.850 value of 21 (call rejected). Note that the 403 response shall not contain the ets RPH header.

<End of VoIP-PSTNGW-SIP-01700>

Priority Treatment Requirements

<VoIP-PSTNGW-Priority-00100>

The PSTNGW shall provide priority treatment to NS/EP SIP signaling messages.

<End of VoIP-PSTNGW-Priority-00100>

NOTE: The intent of the above requirement is to discard recognized NS/EP packets and messages last, and to provide priority processing of the NS/EP packets and messages over normal packets and messages where practical.

<VoIP-PSTNGW-Priority-00200>

The PSTNGW shall set the IP header DSCP value to the NS/EP value for NS/EP SIP messages and RTP packets based upon the call being identified as an NS/EP call.

<End of VoIP-PSTNGW-Priority-00200>

<VoIP-PSTNGW-Priority-00300>

The PSTNGW shall provide priority treatment for NS/EP related SIP or RTP packets at the IP layer (layer 3). This is implemented as a separate queuing treatment from POTS related SIP and RTP packets.

<End of VoIP-PSTNGW-Priority-00300>

NOTE: The above requirement assumes UDP streams. The implication of the above for SCTP and TCP streams is for further study.

<VoIP-PSTNGW-Priority-00400>

The PSTNGW shall provide a mechanism that will throttle the amount of NS/EP-related requests originating from a source IP address.

<End of VoIP-PSTNGW-Priority-00400>

NOTE: A PSTNGW may be the first media aware VoIP NE in the call flow.

<VoIP-PSTNGW-Priority-00500>

The maximum rate of ETS session requests from an IP address, in terms of sessions attempts per minute, shall be provisionable

<End of VoIP-PSTNGW-Priority-00500>

<VoIP-PSTNGW-Priority-00600>

The maximum number of concurrent ETS sessions from an IP address shall be provisionable.

<End of VoIP-PSTNGW-Priority-00600>

<VoIP-PSTNGW-Priority-00700>

An "ets compliant" device shall create an alert, in real time, of IP addresses exceeding the limits specified above.

<End of VoIP-PSTNGW-Priority-00700>

<VoIP-PSTNGW-Priority-00800>

It is desirable that the "ets compliant" device allow an operator to provision IP addresses exempted from generating an alert when the limits are exceeded.

<End of VoIP-PSTNGW-Priority-00800>

<VoIP-PSTNGW-Priority-00900>

An "ets compliant" device shall allow an operator to either allow or block NS/EP sessions from a given IP address. This requirement may be accomplished through a "black list".

<End of VoIP-PSTNGW-Priority-00900>

<VoIP-PSTNGW-Priority-01000>

The PSTNGW shall exempt all NS/EP related signaling messages and RTP streams from restrictive Network Management Controls.

<End of VoIP-PSTNGW-Priority-01000>

<VoIP-PSTNGW-Priority-01100>

The PSTNGW shall exempt all NS/EP related signaling messages and RTP streams from Automatic Congestion Controls (ACCs).

<End of VoIP-PSTNGW-Priority-01100>

<VoIP-PSTNGW-Priority-01200>

The PSTNGW may only discard non-NS/EP-related signaling messages and RTP streams at the lower levels of Machine Congestion Controls (MCCs). At the highest level of MCC, the PSTNGW may discard NS/EP related signaling messages and RTP streams after all non-NS/EP messages and streams are discarded.

<End of VoIP-PSTNGW-Priority-01200>

<VoIP-PSTNGW-Priority-01300>

Priority treatment shall be provided under overloads up to at least 8x.

<End of VoIP-PSTNGW-Priority-01300>

NOTE: *NS/EP Call Identification*: The following are already addressed by requirements and are provided here for completeness.

1. The BEs shall be able to identify a call as an NS/EP call based upon the following methods:
    a. The dialed number is an NS/EP access number in an SS7 IAM.
    b. The dialed number is an NS/EP access number in a SIP message.
    c. The SS7 IAM CPC = NS/EP.
    d. The SIP RPH = ets.x.

NOTE: *Media Priority Treatment*: The following are already addressed by requirements and are provided here for completeness.

1. The BE shall support the High Capacity Ethernet COS capabilities on the High Capacity Ethernet interface between the BE and the IP router.
2. The BE shall be able to set the Ethernet Frame Header COS parameter on the High Capacity Ethernet interface to the NS/EP value for NS/EP related signaling and RTP IP packets. The NS/EP COS value will have the highest media priority treatment on the High Capacity interface. (Network control may have a higher priority.)
3. The BEs shall be able to set the High Capacity NS/EP COS value based upon the NS/EP DSCP values.
4. The BEs shall support DiffServ (RFCs 2475, 3270).
5. BEs with TDM trunks and trunk groups shall have a Trunk Queuing (Office Wide Queuing) capability on the TDM circuits.

NOTE: *Signaling Mapping*: The following are already addressed by requirements and are provided here for completeness.

1. The Border Elements shall be able to set the SIP RPH based upon the ETS access numbers.

2. The BEs that have SS7/TDM on one side and SIP on the other shall set the outgoing SIP RPH ets.x namespace based upon the incoming SS7 IAM CPC = NS/EP.
3. The BEs that have SS7/TDM on one side and SIP on the other shall set the outgoing SIP RPH wps.x namespace based upon the incoming SS7 IAM. If the SS7 includes a Precedence parameter with a wps MLPP Service Domain, and precedence level = y, the BE shall also set the wps.y namespace.
4. The BEs that have SS7/TDM on one side and SIP on the other shall set the outgoing SS7 ISUP IAM CPC to NS/EP based upon the incoming SIP RPH ets.x namespace being present.
5. The BEs that have SS7/TDM on one side and SIP on the other shall set the outgoing SS7 ISUP IAM Precedence parameter with the wps MLPP Service Domain set to the wps y domain, and the precedence level set to y based upon the incoming SIP RPH wps.y namespace being present.

NOTE: *Signaling Priority Treatment*: The following are already addressed by requirements and are provided here for completeness.

1. The BE shall be able to give priority treatment to NS/EP signaling messages.
2. The BEs that support SS7 shall set the IAM MTP priority level to 1 for NS/EP identified calls.

NOTE: *Vertical Interface*: The following are already addressed by requirements and are provided here for completeness.

1. The BE shall be able to set the IP header DSCP value to the NS/EP values for both NS/EP RTP and NS/EP SIP message IP packets based upon the call being identified as an NS/EP call. Different DSCP values may be used for NS/EP signaling and NS/EP bearer.
2. The BE shall be able to set a layer 2 (e.g., COS bits in GIGE) value to the NS/EP values for both NS/EP RTP and NS/EP SIP message IP packets based upon the call being identified as an NS/EP call. Different values may be used for NS/EP signaling and NS/EP bearer.

NOTE: *Network Management Exemptions*: The following are already addressed by requirements and are provided here for completeness.

1. The BEs that support a TDM side as well as a SIP side shall be able to exempt NS/EP identified calls from restrictive TDM NM controls. These include: CANT, CANF, RR, DRE, STR DR, other.
2. The BEs shall exempt NS/EP identified calls from any restrictive VoIP network management controls. Since VoIP NM controls are not well defined at this point, this general requirement shall suffice.
3. The BEs shall be able to measure NS/EP calls that are exempt from the various TDM and VoIP controls and send the data to a performance monitoring application.

6.2.1 Operational Measurements

The objectives of the operational measurements are:

1. To allow calculation of the successful call completion percentage for NS/EP calls/sessions at a given NE.
2. To allow analysis of the conditions (such as errors) at a given NE that caused NS/EP calls/sessions to fail, so that corrective action can be taken.

The purpose of the operational measurements is to provide the NCS with a snapshot into the health of all ETS provisioned switches. Peg counts provide a uniform way to perform this analysis across the thousands of switches anticipated in the NGN. A peg count may indicate further analyses of log files are required to understand the problem associated with a switch.

6.2.2 Successful Call Completion Operational Measurements

The successful call completion percentage can be derived from the following calculations:

Successful call completion percentage = (successful calls) / (call attempts)

Successful calls = (answered calls) + (caller abandoned calls) + (called user busy calls)

Call attempts = (call requests received) – (call requests reject with a 403 error)

The following measurements allow the above calculations to be made.

<VoIP-PSTNGW-Measurements-00100>

The PSTNGW BEs shall count the number of Incoming NS/EP (ETS) call/session requests received.

<End of VoIP-PSTNGW-Measurements-00100>

NOTE: This measurement would be pegged upon receipt of a request in the TDM domain or the IP domain. The request may contain an ETS-DN and no NS/EP marking, or some indication that the request is an NS/EP request (e.g., an RPH with an ets namespace or a CPC="NSEP call" parameter in an IAM). Note that <VoIP-PSTNGW-Measurements-00100> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request.

<VoIP-PSTNGW-Measurements-00200>

The PSTNGW BEs shall count the number of completed NS/EP calls/sessions that are answered.

<End of VoIP-PSTNGW-Measurements-00200>

NOTE: The call may be completed in either the IP domain or the TDM domain. In the TDM domain, this measurement would be pegged after receipt of an SS7 answer message (ANM). In the IP domain, this measurement would be pegged after receipt of the 200 message associated with the initial INVITE.

<VoIP-PSTNGW-Measurements-00300>

The PSTNGW BEs shall count the number of completed NS/EP call/session requests that are unanswered due to a "caller abandoned" ("ring-no-answer") condition.

<End of VoIP-PSTNGW-Measurements-00300>

NOTE: This measurement would be pegged upon receipt of a message in the TDM domain or the IP domain. In the TDM domain, the measurement would be pegged after receipt of an SS7 release message (REL) with a cause code of 18 (no user responding) or 19 (no answer from user). In the IP domain, this measurement would be pegged after receipt of a CANCEL message associated with the initial INVITE.

<VoIP-PSTNGW-Measurements-00400>

The PSTNGW BEs shall count the number of completed NS/EP call/session requests that are unanswered due to a "user busy" condition.

<End of VoIP-PSTNGW-Measurements-00400>

NOTE: This measurement would be pegged upon receipt of a message in the TDM domain or the IP domain. In the TDM domain, this measurement would be pegged after receipt of an SS7 release message (REL) with a cause code of 17 (user busy). In the IP domain, this measurement would be pegged after receipt of a 486 or a 600 message associated with the initial INVITE.

<VoIP-PSTNGW-Measurements-00500>

The PSTNGW BEs shall count the number of incomplete NS/EP call/session requests due to <VoIP-PSTNGW-SIP-01700>.

<End of VoIP-PSTNGW-Measurements-00500>

NOTE: In the IP domain, this measurement would be pegged after generation of a 403 message associated with the initial INVITE. There is no corresponding measurement associated with the TDM side.

6.2.1.2 Operational Measurements Identifying NE Conditions that Impact NS/EP Calls

The number of secondary routes that were attempted by the NE to complete an NS/EP call can be derived from the following calculation:

Secondary routes attempted = (call requests sent) – (call requests received) + (call requests rejected)

Call requests received were counted in <VoIP-PSTNGW-Measurements-00100>, and call requests rejected were counted in <VoIP-PSTNGW-Measurements-00500>. In addition to these measurements, the following measurement allows the above calculation to be made.

<VoIP-PSTNGW-Measurements-00600>

The PSTNGW BEs shall count the number of Outgoing NS/EP (ETS) call/session requests sent.

<End of VoIP-PSTNGW-Measurements-00600>

NOTE: This measurement would be pegged upon transmission of a message in the TDM domain (IAM) or the IP domain (INVITE). The request being sent should contain the appropriate NS/EP markings. Note that <VoIP-PSTNGW-Measurements-00600> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request

The number of “error” messages generated by the PSTNGW can be calculated as follows:

Error messages generated = (error messages sent) – (error messages received) + (secondary routes attempted)

(secondary routes attempted) was calculated above. <VoIP-PSTNGW-Measurements-00700> and <VoIP-PSTNGW-Measurements-00800> capture the error messages sent and received.

<VoIP-PSTNGW-Measurements-00700>

The PSTNGW BEs shall count the number of NS/EP call/session “error” messages received from downstream network NEs.

<End of VoIP-PSTNGW-Measurements-00700>

NOTE: In the TDM domain, the received message is typically a REL with a cause code other than 17, 18, or 19. In the IP domain, the received message is typically a 4xx (except 486), 5xx, or 6xx (except 600) response.

<VoIP-PSTNGW-Measurements-00800>

The PSTNGW BEs shall count the number of NS/EP call/session “error” messages sent to upstream NEs.

<End of VoIP-PSTNGW-Measurements-00800>

NOTE: This measurement would be pegged upon transmission of an error message in the TDM domain (typically a REL with a cause code other than 17, 18, or 19) or in the IP domain [typically a 4xx (except 403 and 486), 5xx, or 6xx (except 600) response].

<VoIP-PSTNGW-Measurements-00900>

The PSTNGW BEs shall count the number of messages sent to a log file associated with requirements <VoIP-GEN-SIP-01200> (417 and 420 errors), <VoIP-GEN-SIP-01400> through <VoIP-GEN-SIP-02000> (400 errors), and <VoIP-GEN- SIP-02500> (NS/EP messages received by an NE for a non-NS/EP session).

<End of VoIP-PSTNGW-Measurements-00900>

NOTE: The above measurement counts the number of NS/EP requests where an error occurred due to the RPH header. Detailed information is captured in a log file for further analysis. This measurement only applies to the IP domain.

<VoIP-PSTNGW-Measurements-01000> through <VoIP-PSTNGW-Measurements-01200> count the number of NS/EP call/session requests that attempt to queue for media resources. Note that queuing may occur in either the TDM domain or in the IP domain.

<VoIP-PSTNGW-Measurements-01000>

The PSTNGW BEs shall count the number of NS/EP call/session requests for which queuing for media resources is attempted.

<End of VoIP-PSTNGW-Measurements-01000>

NOTE: This measurement is pegged for both the IP domain and for the TDM domain.

<VoIP-PSTNGW-Measurements-01100>

The PSTNGW BEs shall count the number of NS/EP call/session requests which are queued for media resources and time out of the queue.

<End of VoIP-PSTNGW-Measurements-01100>

NOTE: This measurement is pegged for both the IP domain and for the TDM domain. In the IP domain, this measurement would be pegged after generating a 408 response with a Q.850 value of 102.

<VoIP-PSTNGW-Measurements-01200>

The PSTNGW BEs shall count the number of NS/EP call/session requests for which media resource queuing is attempted, but the call/session cannot be queued due to the queue being filled to its maximum capacity.

<End of VoIP-PSTNGW-Measurements-01200>

NOTE: This measurement is pegged for both the IP domain and for the TDM domain. In the IP domain, this measurement would be pegged after generating a 503 response for this condition. (Note that a 503 response will also be generated for conditions other than a full queue.)

<VoIP-PSTNGW-Measurements-01300>

The PSTNGW BEs shall count the number of NS/EP calls exempted from VoIP NM Controls, including ACC.

<End of VoIP-PSTNGW-Measurements-01300>

NOTE: This measurement applies to both the TDM domain and to the IP domain.

<VoIP-PSTNGW-Measurements-01400>

The PSTNGW BEs shall count the total number of NS/EP call/session requests rejected due to call admission controls and other NS/EP throttling mechanisms.

<End of VoIP-PSTNGW-Measurements-01400>

NOTE: This measurement only applies to the IP domain.

<VoIP-PSTNGW-Measurements-01500>

All measurement counts shall be for a provisionable period of time. The default count interval shall be 30 minutes, with a maximum provisionable value of 24 hours.

<End of VoIP-PSTNGW-Measurements-01500>

6.3 IP Border Element Requirements (Access/Interconnection SBC)

Assumption:

1. All **VoIP-GEN-SIP** requirements defined in 6.1 are supported.

<VoIP-IPBE-SIP-00100>

Upon receipt of an SIP INVITE with To Header =ETS DN, for subsequent SIP messages in response to the INVITE, the IPBE shall create the RPH with an ets namespace. The ets value shall have a default value; the default value shall be configurable.

An ETS DN is a provisionable parameter provisioned against dialable numbers in the appropriate translation tables.

<End of VoIP-IPBE-SIP-00100>

<VoIP-IPBE-SIP-00200>

VoIP IPBEs that have implemented VoIP ETS priority treatment features shall trigger these features based upon the receipt of RPH=ets.x and/or a SIP INVITE with To header = ETS-DN.

An IPBE NE shall recognize an incoming ETS call by:

a. The presence of a valid RPH of ets.x; and/or
b. The presence of an ETS-DN (e.g., 710-NXX-XXXX).

At a minimum, 10 ETS-DNs shall be configurable or provisionable. A DN can be a partial string (e.g., 710) or a complete number (e.g., 710 NCS GETS). Matching of the ETS-DNs shall start with the leading digit of the received called number.

<End of VoIP-IPBE-SIP-00200>

<VoIP-IPBE-SIP-00300>

A request received with an ETS-DN is assumed to be an ETS (NS/EP) call, and the device shall provide NS/EP treatment to the call. The IPBE may set or modify the values associated with the ets and wps namespaces, based on provisionable parameters.

<End of VoIP-IPBE-SIP-00300>

<VoIP-IPBE-SIP-00400>

The IPBE shall be configurable to accept or reject a SIP INVITE with the ets RPH and a normal (non-ETS-DN) destination number on a per interface basis. The default shall be to reject the SIP INVITE. It is recommended that the reject be a 403 (Forbidden) response with two reason header fields: (1) a 417 (RPH header) code for SIP; and (2) a Reason Header Q.850 value of 21 (call rejected). Note that the 403 response shall not contain the ets RPH header.

<End of VoIP-IPBE-SIP-00400>

Priority Treatment Requirements

<VoIP-IPBE-Priority-00100>

The IPBE shall provide priority treatment to ETS SIP signaling messages.

<End of VoIP-IPBE-Priority-00100>

NOTE: The intent of the above requirement is to discard recognized NS/EP packets and messages last, and to provide priority processing of the NS/EP packets and messages over normal packets and messages where practical.

<VoIP-IPBE-Priority-00200>

The IPBE shall set the IP header DSCP value to the ETS values for NS/EP SIP messages and RTP packets based upon the call being identified as an ETS call.

<End of VoIP-IPBE-Priority-00200>

<VoIP-IPBE-Priority-00300>

The IPBE shall provide priority treatment for ETS related SIP or RTP packets at the IP layer (layer 3). This is implemented as a separate queuing treatment from POTS related SIP and RTP packets.

<End of VoIP-IPBE-Priority-00300>

NOTE: The above requirement assumes UDP streams. The implication of the above for SCTP and TCP streams is for further study.

<VoIP-IPBE-Priority-00400>

The IPBE shall provide a mechanism that will throttle the amount of ETS-related requests originating from a source IP address.

<End of VoIP-IPBE-Priority-00400>

<VoIP-IPBE-Priority-00500>

The maximum rate of ETS session requests from an IP address, in terms of sessions attempts per minute, shall be provisionable.

<End of VoIP-IPBE-Priority-00500>

<VoIP-IPBE-Priority-00600>

The maximum number of concurrent ETS sessions from an IP address shall be provisionable.

<End of VoIP-IPBE-Priority-00600>

<VoIP-IPBE-Priority-00700>

An "ets compliant" device shall create an alert, in real time, of IP addresses exceeding the limits specified above.

<End of VoIP-IPBE-Priority-00700>

<VoIP-IPBE-Priority-00800>

It is desirable that the "ets compliant" device allow an operator to provision IP addresses exempted from generating an alert when the limits are exceeded.

<End of VoIP-IPBE-Priority-00800>

<VoIP-IPBE-Priority-00900>

An "ets compliant" device shall allow an operator to either allow or block ETS sessions from a given IP address. This requirement may be accomplished through a "black list".

<End of VoIP-IPBE-Priority-00900>

<VoIP-IPBE-Priority-01000>

All ETS related signaling messages and RTP streams shall be exempt from restrictive Network Management Controls.

<End of VoIP-IPBE-Priority-01000>

<VoIP-IPBE-Priority-01100>

All ETS related signaling messages and RTP streams shall be exempt from Automatic Congestion Controls (ACCs).

<End of VoIP-IPBE-Priority-01100>

<VoIP-IPBE-Priority-01200>

The IPBE may only discard non-NS/EP-related signaling messages and RTP streams at the lower levels of Machine Congestion Controls (MCCs). At the highest level of MCC, the IPBE may discard NS/EP related signaling messages and RTP streams after all non-NS/EP messages and streams are discarded.

<End of VoIP-IPBE-Priority-01200>

<VoIP-IPBE-Priority-01300>

The IPBE shall exempt ETS identified calls from VoIP CAC/QoS policy restrictions (PDP or PEP).

The IPBE shall provide priority exemption from CAC/QoS congestion controls, both internal and external.

<End of VoIP-IPBE-Priority-01300>

<VoIP-IPBE-Priority-01400>

Priority treatment shall be provided under overloads up to at least 8x.

<End of VoIP-IPBE-Priority-01400>

6.3.1 Measurements

6.3.1.1 Successful Call Completion Operational Measurements

The successful call completion percentage can be derived from the following calculations:

Successful call completion percentage = (successful calls) / (call attempts)

Successful calls = (answered calls) + (caller abandoned calls) + (called user busy calls)

Call attempts = (call requests received) – (call requests reject with a 403 error)

The following measurements allow the above calculations to be made.

<VoIP-IPBE-Measurements-00100>

The IPBE shall count the number of Incoming NS/EP (ETS) call/session requests received.

<End of VoIP-IPBE-Measurements-00100>

NOTE: An ETS request may contain an ETS-DN and no NS/EP marking, or an RPH with an ets namespace. Note that <VoIP-IPBE-Measurements-00100> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request.

<VoIP-IPBE-Measurements-00200>

The IPBE shall count the number of completed NS/EP calls/sessions that are answered.

<End of VoIP-IPBE-Measurements-00200>

NOTE: In the IP domain, this measurement would be pegged after receipt of the 200 message associated with the initial INVITE.

<VoIP-IPBE-Measurements-00300>

The IPBE shall count the number of completed NS/EP call/session requests that are unanswered due to a "caller abandoned" ("ring-no-answer") condition.

<End of VoIP-IPBE-Measurements-00300>

NOTE: In the IP domain, this measurement would be pegged after receipt of a CANCEL message associated with the initial INVITE.

<VoIP-IPBE-Measurements-00400>

The IPBE shall count the number of completed NS/EP call/session requests that are unanswered due to a "user busy" condition.

<End of VoIP-IPBE-Measurements-00400>

NOTE: In the IP domain, this measurement would be pegged after receipt of a 486 or a 600 message associated with the initial INVITE.

<VoIP-IPBE-Measurements-00500>

The IPBE shall count the number of incomplete NS/EP call/session requests due to <VoIP-IPBE-SIP-00400>.

<End of VoIP-IPBE-Measurements-00500>

NOTE: In the IP domain, this measurement would be pegged after generation of a 403 message associated with the initial INVITE.

6.3.1.2 Operational Measurements Identifying NE Conditions that Impact NS/EP Calls

The number of secondary routes that were attempted by the NE to complete an NS/EP call can be derived from the following calculation:

Secondary routes attempted = (call requests sent) – (call requests received) + (call requests rejected)

Call requests received were counted in <VoIP-IPBE-Measurements-00100>, and call requests rejected were counted in <VoIP-IPBE-Measurements-00500>**.** In addition to these measurements, the following measurement allows the above calculation to be made.

<VoIP-IPBE-Measurements-00600>

The IPBE shall count the number of Outgoing NS/EP (ETS) call/session requests sent.

<End of VoIP-IPBE-Measurements-00600>

NOTE: A request should contain the appropriate NS/EP markings. Note that <VoIP-IPBE-Measurements-00600> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request

The number of "error" messages generated by the IPBE can be calculated as follows:

Error messages generated = (error messages sent) – (error messages received) + (secondary routes attempted)

(secondary routes attempted) was calculated above. <VoIP-IPBE-Measurements-00700> and <VoIP-IPBE-Measurements-00800> capture the error messages sent and received.

<VoIP-IPBE-Measurements-00700>

The IPBE shall count the number of NS/EP call/session "error" messages received from downstream network NEs.

<End of VoIP-IPBE-Measurements-00700>

NOTE: In the IP domain, the received message is typically a 4xx (except 486), 5xx, or 6xx (except 600) response.

<VoIP-IPBE-Measurements-00800>

The IPBE shall count the number of NS/EP call/session "error" messages sent to upstream NEs.

<End of VoIP-IPBE-Measurements-00800>

NOTE: This measurement would be pegged upon transmission of an error message in the IP domain (typically a 4xx (except 403 and 486), 5xx, or 6xx (except 600) response).

<VoIP-IPBE-Measurements-00900>

The IPBE shall count the number of messages sent to a log file associated with requirements <VoIP-GEN-SIP-01200> (417 and 420 errors), <VoIP-GEN-SIP-01400> through <VoIP-GEN-SIP-02000> (400 errors), and <VoIP-GEN- SIP-02500> (NS/EP messages received by an NE for a non-NS/EP session).

<End of VoIP-IPBE-Measurements-0900>

NOTE: The above measurement counts the number of NS/EP requests where an error occurred due to the RPH header. Detailed information is captured in a log file for further analysis.

<VoIP-IPBE-Measurements-01000> through <VoIP-IPBE-Measurements-01200> count the number of NS/EP call/session requests that attempt to queue for media resources.

<VoIP-IPBE-Measurements-01000>

The IPBE shall count the number of NS/EP call/session requests for which media resource queuing is attempted.

<End of VoIP-IPBE-Measurements-01000>

<VoIP-IPBE-Measurements-01100>

The IPBE shall count the number of NS/EP call/session requests which are queued for media resources and time out of the queue.

<End of VoIP-IPBE-Measurements-01100>

NOTE: In the IP domain, this measurement would be pegged after generating a 408 response with a Q.850 value of 102.

<VoIP-IPBE-Measurements-01200>

The IPBE shall count the number of NS/EP call/session requests for which media resource queuing is attempted, but the call/session cannot be queued due to the queue being filled to its maximum capacity.

<End of VoIP-IPBE-Measurements-01200>

NOTE: In the IP domain, this measurement would be pegged after generating a 503 response for this condition. (Note that a 503 response will also be generated for conditions other than a full queue.)

<VoIP-IPBE-Measurements-01300>

The IPBE shall count the number of NS/EP calls exempted from VoIP NM Controls, including ACC.

<End of VoIP-IPBE-Measurements-01300>

<VoIP-IPBE-Measurements-01400>

The IPBE shall count the total number of NS/EP call/session requests rejected due to call admission controls and other NS/EP throttling mechanisms.

<End of VoIP-IPBE-Measurements-01400>

<VoIP-IPBE-Measurements-01500>

All measurement counts shall be for a provisionable period of time. The default count interval shall be 30 minutes, with a maximum provisionable value of 24 hours.

<End of VoIP-IPBE-Measurements-01500>

6.4 Media Server

Assumptions:

1. The MS shall support RFC 4733 for all supported codecs, including G.711.
2. A MS is considered “dedicated” if it is only used for ETS service processing. A MS is considered “shared” if it is used for ETS service processing and for purposes other than ETS (e.g., network announcements,

other service processing). Whether a MS is shared or dedicated is dependent on a given network's configuration.

3. All requirements in this section apply to both dedicated and shared MSs.
4. The MS is used to collect PIN and DN information associated with NS/EP calls, and drops out of the signaling path after transmission of this information to the AS.
5. The MS will not receive an INVITE with an ETS-DN, but may receive an INVITE with an ets RPH.

<MS-SIP-00100>

An MS shall implement the set of General SIP requirements specified in this document.

<End of MS-SIP-00100>

<MS-SIP-00200>

An MS that has implemented ETS priority treatment features shall trigger these features based upon the receipt of RPH=ets.x.

<End of MS-SIP-00200>

<MS-SIP-00300>

If an MS cannot support a call request due to an incompatibility between the offered and available codecs, the MS shall generate a 488 response (Not Acceptable Here). The response shall include a Reason Header Q.850 value of 29 (Facility Rejected).

<End of MS-SIP-00300>

Priority Treatment Requirements

<MS-Priority-00100>

The MS shall support the marking of DiffServ per RFCs 2475 and 3270.

<End of MS-Priority-00100>

<MS-Priority-00200>

The MS shall provide priority treatment to ETS SIP and HTTP signaling messages.

<End of MS-Priority-00200>

NOTE: The intent of the above requirement is to discard recognized NS/EP packets and messages last, and to provide priority processing of the NS/EP packets and messages over normal packets and messages where practical.

<MS-Priority-00300>

The MS shall set the IP header DSCP value to the ETS values for ETS SIP and HTTP messages, and RTP packets based upon the call being identified as an ETS call.

<End of MS-Priority-00300>

<MS-Priority-00400>

The MS shall provide priority treatment for ETS related SIP, HTTP, and RTP packets at the IP layer 3. This is implemented as a separate queuing treatment from POTS-related SIP and RTP packets.

<End of MS-Priority-00400>

NOTE: The above requirement assumes UDP streams. The implication of the above for SCTP and TCP streams is for further study.

<MS-Priority-00500>

The MS shall provide a mechanism that will throttle the amount of ETS-related requests originating from a source IP address.

<End of MS-Priority-00500>

<MS-Priority-00600>

All ETS-related signaling messages and RTP streams shall be exempt from any restrictive Network Management Controls.

<End of MS-Priority-00600>

<MS-Priority-00700>

All ETS-related signaling messages and RTP streams shall be exempt from any Automatic Congestion Controls (ACCs).

<End of MS-Priority-00700>

<MS-Priority-00800>

The MS may only discard non-NS/EP-related signaling messages and RTP streams at the lower levels of Machine Congestion Controls (MCCs). At the highest level of MCC, the MS may discard NS/EP related signaling messages and RTP streams after all non-NS/EP messages and streams are discarded.

<End of MS-Priority-00800>

<MS-Priority-00900>

The IMS MS should be able to provide additional routing based upon receipt of an INVITE with an RPH=ets.x, if so provisioned.

<End of MS-Priority-00900>

<MS-Priority-01000>

If any CAC/QoS policy restrictions (PDP or PEP) are active, the MS shall exempt ETS identified calls from such restrictions. The MS shall provide priority exemption from CAC/QoS congestion controls, both internal and external.

<End of MS-Priority-01000>

<MS-Priority-01100>

The MS shall set the IP header DSCP value to the ETS values for ETS-call-related non-SIP messages, such as Diameter and HTTP, based upon the call being identified as an ETS call.

<End of MS-Priority-01100>

<MS-Priority-01200>

Priority treatment shall be provided under overloads up to at least 8x.

<End of MS-Priority-01200>

<MS-Priority-01300>

All MS applicable requirements shall apply to the MS operating in both a VoIP and in an IMS environment.

<End of MS-Priority-01300>

6.4.1 Measurements

The MS Measurements are based on the following assumptions:

- The MS drops out of the call path after the PIN and DN are transferred to the AS and the AS terminates the session. Thus, the MS can experience a resource unavailable condition, but not a user busy condition.
- The MS is an endpoint and does not forward INVITEs to another MS or AS for processing.

6.4.1.1 Successful Call Completion Operational Measurements

The successful call completion percentage can be derived from the following calculations:

Successful call completion percentage = (successful calls) / (call attempts)

Successful calls = (answered calls) + (caller abandoned calls)

Call attempts = (call requests received)

The successful call completion percentage above refers to calls completed to the MS, and not to calls completed to the called party.

The following measurements allow the above calculations to be made.

<MS-Measurements-00100>

The MS shall count the number of Incoming NS/EP (ETS) call/session requests received.

<End of MS-Measurements-00100>

NOTE: An ETS request will contain an RPH with an ets namespace. Note that <MS-Measurements-00100> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request.

<MS-Measurements-00200>

The MS shall count the number of completed NS/EP calls/sessions that are answered.

<End of MS-Measurements-00200>

NOTE: In the IP domain, this measurement would be pegged after receipt of the ACK message the MS receives to the 200 message the MS sent associated with the initial ets INVITE.

<MS-Measurements-00300>

The MS shall count the number of completed NS/EP call/session requests that are unanswered due to a "caller abandoned" ("ring-no-answer") condition.

<End of MS-Measurements-00300>

NOTE: In the IP domain, this measurement would be pegged after receipt of a CANCEL message associated with the initial INVITE.

6.4.1.2 Operational Measurements Identifying NE Conditions that Impact NS/EP Calls

<MS-Measurements-00400>

The MS shall count the number of NS/EP call/session "error" messages sent to upstream NEs.

<End of MS-Measurements-00400>

NOTE: This measurement would be pegged upon transmission of an error message in the IP domain (typically a 4xx, 5xx, or 6xx response).

<MS-Measurements-00500>

The MS shall count the number of messages sent to a log file associated with requirements <VoIP-GEN-SIP-01400> through <VoIP-GEN-SIP-02000> (400 response messages).

<End of MS-Measurements-00500>

NOTE: The above measurement counts the number of NS/EP requests received where the ets and/or wps values in the Resource-Priority header are determined to be invalid.

<MS-Measurements-00600> through <MS-Measurements-00800> count the number of NS/EP call/session requests that attempt to queue for media resources.

<MS-Measurements-00600>

The MS shall count the number of NS/EP call/session requests for which media resource queuing is attempted.

<End of MS-Measurements-0060>

<MS-Measurements-00700>

The MS shall count the number of NS/EP call/session requests which are queued for media resources and time out of the queue.

<End of MS-Measurements-00700>

NOTE: In the IP domain, this measurement would be pegged after generating a 408 response with a Q.850 value of 102.

<MS-Measurements-00800>

The MS shall count the number of NS/EP call/session requests for which media resource queuing is attempted, but the call/session cannot be queued due to the queue being filled to its maximum capacity.

<End of MS-Measurements-00800>

NOTE: In the IP domain, this measurement would be pegged after generating a 503 response for this condition. (Note that a 503 response will also be generated for conditions other than a full queue.)

<MS-Measurements-00900>

The MS shall count the number of NS/EP calls exempted from VoIP NM Controls, including ACC.

<End of MS-Measurements-00900>

<MS-Measurements-01000>

The MS shall count the total number of NS/EP call/session requests rejected due to NS/EP throttling mechanisms.

<End of MS-Measurements-01000>

<MS-Measurements-01100>

All measurement counts shall be for a provisionable period of time. The default count interval shall be 30 minutes, with a maximum provisionable value of 24 hours.

<End of MS-Measurements-01100>

6.5 Application Server

Assumptions:

The requirements listed here assume a shared AS implementation. For a stand-alone ETS AS implementation, some of the priority treatment requirements do not need to be developed.

<VoIP-AS-SIP-00100>

A VoIP AS shall implement the set of General SIP requirements specified in this document.

<End of VoIP-AS-SIP-00100>

<VoIP-AS-SIP-00200>

A VoIP AS that has implemented VoIP ETS priority treatment features shall trigger these features based upon the receipt of RPH=ets.x

<End of VoIP-AS-SIP-00200>

Priority Treatment Requirements

<VoIP-AS-Priority-00100>

The VoIP AS shall support the marking of DiffServ per RFCs 2475 and 3270.

<End of VoIP-AS-Priority-00100>

<VoIP-AS-Priority-00200>

The VoIP AS shall provide priority treatment to ETS SIP signaling messages.

<End of VoIP-AS-Priority-00200>

NOTE: The intent of the above requirement is to discard recognized NS/EP packets and messages last, and to provide priority processing of the NS/EP packets and messages over normal packets and messages where practical.

<VoIP-AS-Priority-00300>

The VoIP AS shall set the IP header DSCP value to the ETS values for ETS SIP message packets based upon the call being identified as an ETS call.

<End of VoIP-AS-Priority-00300>

<VoIP-AS-Priority-00400>

The VoIP AS shall provide priority treatment for ETS related SIP packets at the IP layer 3. This is implemented as a separate queuing treatment from POTS/VoIP related SIP packets.

<End of VoIP-AS-Priority-00400>

NOTE: The above requirement assumes UDP streams. The implication of the above for SCTP and TCP streams is for further study.

<VoIP-AS-Priority-00500>

The VoIP AS shall provide a mechanism that will throttle the amount of ETS-related requests originating from a source IP address.

<End of VoIP-AS-Priority-00500>

<VoIP-AS-Priority-00600>

All ETS-related signaling message streams shall be exempt from any restrictive Network Management Controls.

<End of VoIP-AS-Priority-00600>

<VoIP-AS-Priority-00700>

All ETS-related signaling message streams shall be exempt from any Automatic Congestion Controls (ACCs).

<End of VoIP-AS-Priority-00700>

<VoIP-AS-Priority-00800>

The AS may only discard non-NS/EP-related signaling messages and streams at the lower levels of Machine Congestion Controls (MCCs). At the highest level of MCC, the AS may discard NS/EP related signaling messages and streams after all non-NS/EP messages and streams are discarded.

<End of VoIP-AS-Priority-00800>

<VoIP-AS-Priority-00900>

The AS shall set the Ethernet COS value to the ETS values for ETS SIP message packets based upon the call being identified as an ETS call.

<End of VoIP-AS-Priority-00900>

<VoIP-AS-Priority-01000>

The AS shall set the IP header DSCP value to the ETS values for ETS-call-related non-SIP messages, such as Diameter and HTTP, based upon the call being identified as an ETS call.

<End of VoIP-AS-Priority-01000>

<VoIP-AS-Priority-01100>

The AS shall provide priority treatment for ETS-related SIP packets at the Ethernet layer 2. This is implemented as a separate queuing treatment from POTS/VoIP related SIP Ethernet packets.

<End of VoIP-AS-Priority-01100>

<VoIP-AS-Priority-01200>

All AS applicable requirements shall apply to the AS operating in both a VoIP and in an IMS environment.

<End of VoIP-AS-Priority-01200>

<VoIP-AS-Priority-01300>

Priority treatment shall be provided under overloads up to at least 8x.

<End of VoIP-AS-Priority-01300>

6.5.1 Measurements

The AS Measurements are based on the following assumptions:

- After performing its function, an AS may either drop out of the call signaling path (e.g., an authentication application server) or remain in the call signaling path (e.g., a call admission control application).
- An AS may forward an INVITE to multiple application servers (e.g., the AS is acting as a service broker).
- An AS (e.g., an authentication server) may be able to classify the various types of ETS calls, including ETS-DN calls, ETS calls requiring number translation, and ETS calls identified with a feature code. (Although ETS feature code calls are for further study, the AS may support ETS calls across multiple access and core domains.)
- An AS may communicate with an MS to obtain PIN and DN information. It may then either stay in the call signaling path to the DN, or it may drop from the call.

6.5.1.1 Successful Call Completion Operational Measurements

The successful call completion percentage can be derived from the following calculations:

Successful call completion percentage = (successful calls)/(call attempts)

Successful calls = (INVITEs sent to non-MS NEs)

Call attempts = (call requests received)

The following measurements allow the above calculations to be made.

<VoIP-AS-Measurements-00100>

The AS shall count the number of Incoming NS/EP (ETS) call/session requests received.

<End of VoIP-AS-Measurements-00100>

NOTE: <VoIP-AS-Measurements-00100> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request.

<VoIP-AS-Measurements-00200>

The AS shall count the number of Outgoing NS/EP (ETS) call/session requests sent to non-MS NEs.

<End of VoIP-AS-Measurements-00200>

6.5.1.2 Additional Call Statistics Operational Measurements

NOTE: The following measurements identify the types of ETS calls recognized by the AS for further processing. The AS need only peg those NS/EP message types that it recognizes (e.g., because it is an authentication server).

<VoIP-AS-Measurements-00300>

The AS shall count the number of Incoming NS/EP (ETS) call/session requests received with an ETS-DN and no ETS FC.

<End of VoIP-AS-Measurements-00300>

<VoIP-AS-Measurements-00400>

The AS shall count the number of Incoming NS/EP (ETS) call/session requests received with an ETS FC and no ETS-DN.

<End of VoIP-AS-Measurements-00400>

<VoIP-AS-Measurements-00500>

The AS shall count the number of Incoming NS/EP (ETS) call/session requests received with both an ETS FC and an ETS-DN.

<End of VoIP-AS-Measurements-00500>

<VoIP-AS-Measurements-00600>

The AS shall count the number of Incoming NS/EP (ETS) call/session requests received with a normal DN and no ETS FC.

<End of VoIP-AS-Measurements-00600>

6.5.1.3 Call Authentication Operational Measurements

NOTE: The following measurements identify issues associated with the AS authenticating an ETS call.

<VoIP-AS-Measurements-00700>

The AS shall count the number of NS/EP call/session requests it fails due to an ETS FC authentication problem.

<End of VoIP-AS-Measurements-00700>

NOTE: In this case, the user is not authorized to use an ETS FC.

<VoIP-AS-Measurements-00800>

The AS shall count the number of NS/EP call/session requests it fails due to a PIN authentication problem.

<End of VoIP-AS-Measurements-00800>

NOTE: In this case, the user does not provide an authorized PIN.

<VoIP-AS-Measurements-00900>

The AS shall count the number of NS/EP call/session requests it fails due to a problem with the DN.

<End of VoIP-AS-Measurements-00900>

NOTE: In this case, the user does not provide a DN, or provides an invalid or unreachable DN, or the user is not authorized to access that DN (e.g., international numbers).

<VoIP-AS-Measurements-01000>

The AS shall count the number of NS/EP call/session requests it fails due to an ETS Network Translation DN error.

<End of VoIP-AS-Measurements-01000>

NOTE: In this case, the 710 or 711 number initially provided by the user cannot be translated into a valid DN.

<VoIP-AS-Measurements-01100>

The AS shall count the number of NS/EP call/session requests assumed authenticated due to a "fail open" condition.

<End of VoIP-AS-Measurements-01100>

NOTE: In this case, the database identifying authorized NS/EP users/devices cannot be accessed, so the AS assumes the request is valid.

6.5.1.4 Operational Measurements Identifying NE Conditions that Impact NS/EP Calls

<VoIP-AS-Measurements-01200>

The AS shall count the number of NS/EP call/session "error" messages sent to upstream NEs.

<End of VoIP-AS-Measurements-01200>

NOTE: This measurement would be pegged upon transmission of an error message in the IP domain (typically a 4xx, 5xx, or 6xx response).

<VoIP-AS-Measurements-01300>

The AS shall count the number of messages sent to a log file associated with requirements <VoIP-GEN-SIP-01200> (417 and 420 errors), <VoIP-GEN-SIP-01400> through <VoIP-GEN-SIP-02000> (400 errors), and <VoIP-GEN- SIP-02500> (NS/EP messages received by an NE for a non-NS/EP session).

<End of VoIP-AS-Measurements-01300>

NOTE: The above measurement counts the number of NS/EP requests where an error occurred due to the RPH header. Detailed information is captured in a log file for further analysis.

<VoIP-AS-Measurements-01400> through <VoIP-AS-Measurements-01600> count the number of NS/EP call/session requests that attempt to queue for media resources.

<VoIP-AS-Measurements-01400>

The AS shall count the number of NS/EP call/session requests for which queuing for media resources is attempted.

<End of VoIP-AS-Measurements-01400>

<VoIP-AS-Measurements-01500>

The AS shall count the number of NS/EP call/session requests which are queued for media resources and time out of the queue.

<End of VoIP-AS-Measurements-01500>

NOTE: In the IP domain, this measurement would be pegged after generating a 408 response with a Q.850 value of 102.

<VoIP-AS-Measurements-01600>

The AS shall count the number of NS/EP call/session requests for which media resource queuing is attempted, but the call/session cannot be queued due to the queue being filled to its maximum capacity.

<End of VoIP-AS-Measurements-01600>

NOTE: In the IP domain, this measurement would be pegged after generating a 503 response for this condition. (Note that a 503 response will also be generated for conditions other than a full queue.)

<VoIP-AS-Measurements-01700>

The AS shall count the number of NS/EP calls exempted from VoIP NM Controls, including ACC.

<End of VoIP-AS-Measurements-01700>

<VoIP-AS-Measurements-01800>

The AS shall count the total number of NS/EP call/session requests rejected due to call admission controls and other NS/EP throttling mechanisms.

<End of VoIP-AS-Measurements-01800>

<VoIP-AS-Measurements-01900>

All measurement counts shall be for a provisionable period of time. The default count interval shall be 30 minutes, with a maximum provisionable value of 24 hours.

<End of VoIP-AS-Measurements-01900>

6.6 S/I-CSCF

Assumption:

The S/I-CSCFs are connected to a common backbone edge router by a High Capacity Ethernet interface.

<S/I-CSCF-SIP-00100>

A S/I-CSCF shall implement the set of General SIP requirements specified in this document.

<End of S/I-CSCF-SIP-00100>

<S/I-CSCF-SIP-00200>

A S/I-CSCF that has implemented VoIP ETS priority treatment features shall trigger these features based upon the receipt of RPH=ets.x.

<End of S/I-CSCF-SIP-00200>

Priority Treatment Requirements

<S/I-CSCF-Priority-00100>

The S/I-CSCF shall support the marking of DiffServ per RFCs 2475 and 3270.

<End of S/I-CSCF-Priority-00100>

<S/I-CSCF-Priority-00200>

The S/I-CSCF shall provide priority treatment to ETS SIP signaling messages.

<End of S/I-CSCF-Priority-00200>

NOTE: The intent of the above requirement is to discard recognized NS/EP packets and messages last, and to provide priority processing of the NS/EP packets and messages over normal packets and messages where practical.

<S/I-CSCF-Priority-00300>

The S/I-CSCF shall set the IP header DSCP value to the ETS values for ETS SIP message packets based upon the call being identified as an ETS call.

<End of S/I-CSCF-Priority-00300>

<S/I-CSCF-Priority-00400>

The S/I-CSCF shall provide priority treatment for ETS-related SIP packets at the IP layer 3. This is implemented as a separate queuing treatment from POTS/VoIP related SIP packets.

<End of S/I-CSCF-Priority-00400>

NOTE: The above requirement assumes UDP streams. The implication of the above for SCTP and TCP streams is for further study.

<S/I-CSCF-Priority-00500>

The S/I-CSCF shall provide a mechanism that will throttle the amount of ETS-related requests originating from a source IP address.

<End of S/I-CSCF-Priority-00500>

<S/I-CSCF-Priority-00600>

All ETS related signaling message streams shall be exempt from any restrictive Network Management Controls.

<End of S/I-CSCF-Priority-00600>

<S/I-CSCF-Priority-00700>

All ETS related signaling message streams shall be exempt from any Automatic Congestion Controls (ACCs).

<End of S/I-CSCF-Priority-00700>

<S/I-CSCF-Priority-00800>

The S/I-CSCF may only discard non-NS/EP-related signaling messages and streams at the lower levels of Machine Congestion Controls (MCCs). At the highest level of MCC, the AS may discard NS/EP related signaling messages and streams after all non-NS/EP messages and streams are discarded.

<End of S/I-CSCF-Priority-00800>

<S/I-CSCF-Priority-00900>

The S/I-CSCF shall set the Ethernet COS value to the ETS values for ETS SIP message packets based upon the call being identified as an ETS call.

<End of S/I-CSCF-Priority-00900>

<S/I-CSCF-Priority-01000>

The S/I-CSCF shall set the IP header DSCP value to the ETS values for ETS-call-related non-SIP messages, such as Diameter and HTTP, based upon the call being identified as an ETS call.

<End of S/I-CSCF-Priority-01000>

<S/I-CSCF-Priority-01100>

The S/I-CSCF shall provide priority treatment for ETS related SIP packets at the Ethernet layer 2. This is implemented as a separate queuing treatment from POTS/VoIP related SIP Ethernet packets.

<End of S/I-CSCF-Priority-01100>

<S/I-CSCF-Priority-01200>

Priority treatment shall be provided under overloads up to at least 8x.

<End of S/I-CSCF-Priority-01200>

<S/I-CSCF-Priority-01300>

All S/I-CSCF applicable requirements shall apply to the S/I-CSCF operating in both a VoIP and in an IMS environment.

<End of S/I-CSCF-Priority-01300>

<S/I-CSCF-Priority-01400>

The S/I-CSCF shall provide additional routing based upon receipt of an INVITE with RPH=ets.x, if so provisioned.

<End of S/I-CSCF-Priority -01400>

6.6.1 Measurements

The S/I-CSCF Measurements are based on the following assumption:

- An S/I-CSCF may be able to classify the various types of ETS calls (e.g., via its initial filter criteria), including ETS-DN calls, ETS calls requiring number translation, and ETS calls identified with a feature code.

6.6.1.1 Successful Call Completion Operational Measurements

The successful call completion percentage can be derived from the following calculations:

Successful call completion percentage = (successful calls) / (call attempts)

Successful calls = (answered calls) + (caller abandoned calls) + (called user busy calls)

Call attempts = (call requests received)

The following measurements allow the above calculations to be made.

<S/I-CSCF-Measurements-00100>

The S/I-CSCF shall count the number of Incoming NS/EP (ETS) call/session requests received.

<End of S/I-CSCF-Measurements-00100>

NOTE: The request may contain an ETS-DN and no NS/EP marking, or some indication that the request is an NS/EP request (e.g., an RPH with an ets namespace). Note that <S/I-CSCF-Measurements-00100> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request.

<S/I-CSCF-Measurements-00200>

The S/I-CSCF shall count the number of completed NS/EP calls/sessions that are answered.

<End of S/I-CSCF-Measurements-00200>

NOTE: In the IP domain, this measurement would be pegged after receipt of the 200 message associated with the initial INVITE.

<S/I-CSCF-Measurements-00300>

The S/I-CSCF shall count the number of completed NS/EP call/session requests that are unanswered due to a "caller abandoned" ("ring-no-answer") condition.

<End of S/I-CSCF-Measurements-00300>

NOTE: In the IP domain, this measurement would be pegged after receipt of a CANCEL message associated with the initial INVITE.

<S/I-CSCF-Measurements-00400>

The S/I-CSCF shall count the number of completed NS/EP call/session requests that are unanswered due to a "user busy" condition.

<End of S/I-CSCF-Measurements-00400>

NOTE: In the IP domain, this measurement would be pegged after receipt of a 486 or a 600 message associated with the initial INVITE.

6.6.1.2 Additional Call Statistics Operational Measurements

NOTE: The following measurements identify the types of ETS calls recognized by the S/I-CSCF for further processing. The S/I-CSCF need only peg those NS/EP message types that it recognizes (e.g., because of an initial filter criteria associated with a specific call type).

<S/I-CSCF-Measurements-00500>

The S/I-CSCF shall count the number of Incoming NS/EP (ETS) call/session requests received with an ETS-DN and no ETS FC.

<End of S/I-CSCF-Measurements-00500>

<S/I-CSCF-Measurements-00600>

The S/I-CSCF shall count the number of Incoming NS/EP (ETS) call/session requests received with an ETS FC and no ETS-DN.

<End of S/I-CSCF-Measurements-00600>

<S/I-CSCF-Measurements-00700>

The S/I-CSCF shall count the number of Incoming NS/EP (ETS) call/session requests received with both an ETS FC and an ETS-DN.

<End of S/I-CSCF-Measurements-00700>

<S/I-CSCF-Measurements-00800>

The S/I-CSCF shall count the number of Incoming NS/EP (ETS) call/session requests received with a normal DN and no ETS FC.

<End of S/I-CSCF-Measurements-00800>

6.6.1.3 Operational Measurements Identifying NE Conditions that Impact NS/EP Calls

The number of secondary routes that were attempted by the NE to complete an NS/EP call can be derived from the following calculation:

Secondary routes attempted = (call requests sent) – (call requests received)

Call requests received were counted in <S/I-CSCF-Measurements-00100>. In addition to this measurement, <S/I-CSCF-Measurements-00900> allows the above calculation to be made.

<S/I-CSCF-Measurements-00900>

The S/I-CSCF shall count the number of Outgoing NS/EP (ETS) call/session requests sent.

<End of S/I-CSCF-Measurements-00900>

NOTE: <S/I-CSCF-Measurements-00900> would peg the initial SIP INVITE associated with a session request, but not subsequent (re)INVITEs associated with the same session request.

The number of "error" messages generated by the S/I-CSCF can be calculated as follows:

Error messages generated = (error messages sent) – (error messages received) + (secondary routes attempted)

(secondary routes attempted) was calculated above. <S/I-CSCF-Measurements-01000> and <S/I-CSCF-Measurements-01100> capture the error messages sent and received.

<S/I-CSCF-Measurements-01000>

The S/I-CSCF shall count the number of NS/EP call/session "error" messages received from downstream network NEs.

<End of S/I-CSCF-Measurements-01000>

NOTE: In the IP domain, the received message is typically a 4xx (except 486), 5xx, or 6xx (except 600) response.

<S/I-CSCF-Measurements-01100>

The S/I-CSCF shall count the number of NS/EP call/session "error" messages sent to upstream NEs.

<End of S/I-CSCF-Measurements-01100>

NOTE: This measurement would be pegged upon transmission of an error message in the IP domain (typically a 4xx (except 403 and 486), 5xx, or 6xx (except 600) response).

<S/I-CSCF-Measurements-01200>

The S/I-CSCF shall count the number of messages sent to a log file associated with requirements <VoIP-GEN-SIP-01200> (417 and 420 errors), <VoIP-GEN-SIP-01400> through <VoIP-GEN-SIP-02000> (400 errors), and <VoIP-GEN- SIP-02500> (NS/EP messages received by an NE for a non-NS/EP session).

<End of S/I-CSCF-Measurements-01200>

NOTE: The above measurement counts the number of NS/EP requests where an error occurred due to the RPH header. Detailed information is captured in a log file for further analysis.

NOTE: If the S/I-CSCF is involved in subjecting requests to call admission control, it may queue INVITEs with an ets RPH until the NS/EP call can be admitted. In this case, <S/I-CSCF-Measurements-01300> through <S/I-CSCF-Measurements-01500> count the number of NS/EP call/session requests that attempt to queue for media resources.

<S/I-CSCF-Measurements-01300>

The S/I-CSCF shall count the number of NS/EP call/session requests for which queuing for media resources is attempted.

<End of S/I-CSCF-Measurements-01300>

<S/I-CSCF-Measurements-01400>

The S/I-CSCF shall count the number of NS/EP call/session requests which are queued for media resources and time out of the queue.

<End of S/I-CSCF-Measurements-01400>

NOTE: In the IP domain, this measurement would be pegged after generating a 408 response with a Q.850 value of 102.

<S/I-CSCF-Measurements-01500>

The S/I-CSCF shall count the number of NS/EP call/session requests for which media resource queuing is attempted, but the call/session cannot be queued due to the queue being filled to its maximum capacity.

<End of S/I-CSCF-Measurements-01500>

NOTE: In the IP domain, this measurement would be pegged after generating a 503 response for this condition. (Note that a 503 response will also be generated for conditions other than a full queue.)

<S/I-CSCF-Measurements-01600>

The S/I-CSCF shall count the number of NS/EP calls exempted from VoIP NM Controls, including ACC.

<End of S/I-CSCF-Measurements-01600>

<S/I-CSCF-Measurements-01700>

The S/I-CSCF shall count the total number of NS/EP call/session requests rejected due to call admission controls and other NS/EP throttling mechanisms.

<End of S/I-CSCF-Measurements-01700>

<S/I-CSCF-Measurements-01800>

All measurement counts shall be for a provisionable period of time. The default count interval shall be 30 minutes, with a maximum provisionable value of 24 hours.

<End of S/I-CSCF-Measurements-01800>