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Technical Specification

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS - based PSTN/ISDN Emulation; Stage 3 specification

Modified version for VGW (IAD) connected to accesses of Deutsche Telekom only !

Reference
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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

The present document is a modified version for VGW (IAD) for Deutsche Telekom only and has been produced by the department TE3 of Deutsche Telekom Netzproduktion, Fixed Mobile Engineering Deutschland (in the following named as Deutsche Telekom) and defines the simulation services for IP Multimedia Subsystem (IMS) provided by a SIP User Equipment (UE) via an DSS1 (ISDN) interface (e.g. Integrated Access Device: IAD).

Additional to the present document the modified version of ETSI Technical Specification TS 183 036 [40] (for Deutsche Telekom only) is valid.

NOTE: Text modified due to Deutsche Telekom requirements that is added or deleted is shown as cursive and underlined (example for added text) or cursive and stricken (~~example for stricken text~~).

1 Scope

The present document defines call control protocols and procedures for use in the IMS-based PSTN/ISDN Emulation subsystem based on the Media Gateway Control Protocol (MEGACO), the Session Initiation Protocol (SIP), and the associated Session Description Protocol (SDP).

NOTE: The present document relies on the architectural framework defined in TS 182 012 [3] for IMS-based PES Emulation and may need to be updated once the open issues identified in the present document are resolved.

The present document is applicable to:

- the interface between the User Equipment (UE) and the Call Session Control Function (CSCF);
- ~~*the interface between the Access Gateway Control Function (AGCF) and the Media Gateway Function (MGF);*~~
- ~~*the interface between the Access Gateway Control Function (AGCF) and the Call Session Control Function (CSCF);*~~
- ~~*the interface between the CSCF and any other CSCF;*~~
- ~~*the interface between the CSCF and an Application Server (AS);*~~
- ~~*the interface between the CSCF and the Media Gateway Control Function (MGCF);*~~
- ~~*the interface between the S-CSCF and the Multimedia Resource Function Controller (MRFC);*~~
- ~~*the interface between the CSCF and the Breakout Gateway Control Function (BGCF);*~~
- ~~*the interface between the BGCF and the MGCF;*~~
- ~~*the interface between the BGCF and any other BGCF;*~~
- ~~*the interface between the CSCF and an external Multimedia IP network;*~~
- ~~*the interface between the CSCF and the IBCF.*~~

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
  - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
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2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ETSI ES 282 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Functional Architecture".

[2] ETSI TS 182 006: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Subsystem (IMS); Stage 2 description [3GPP TS 23.506 Release 8, modified]".

[3] ETSI TS 182 012: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation Sub-system (PES); Functional architecture".

[4] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 [Release 7], modified]".

[5] ETSI ES 283 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); H.248 Profile for controlling Access and Residential Gateways".

[6] ETSI TS 183 028: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common Basic Communication procedures; Protocol specification".

[7] ETSI TS 183 047: TISPAN NGN "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN IMS Supplementary Services; Advice of Charge (AoC)".

[8] ETSI TS 183 010: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol; Communication HOLD (HOLD) PSTN/ISDN simulation services; Protocol specification".

[9] ETSI TS 183 004: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Communication Diversion (CDIV); Protocol specification".

[10] ETSI ES 200 659-3: "Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 3: Data link message and parameter codings".

[11] ETSI EG 201 973-2: "Access and Terminals (AT); Public Switched Telephone Network; Support of legacy terminals by Broadband IP networks and equipment; Part 2: Analogue PSTN terminals".

[12] ETSI ETS 300 738: "Human Factors (HF); Minimum Man-Machine Interface (MMI) to public network based supplementary services".

[13] ITU-T Recommendation H.248.23: "Gateway control protocol: Enhanced Alerting packages".

[14] ITU-T Recommendation H.248.26: "Gateway control protocol: Enhanced analog lines packages".

[15] IETF draft-ietf-sipping-config-framework-15: "A Framework for Session Initiation Protocol User Agent Profile Delivery".

[16] IETF RFC 4240: "Basic Network Media Services with SIP".

[17] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[18] IETF RFC 3842: "A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)".

[19] IETF RFC 3966: "The tel URI for Telephone Numbers".

[20] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".

[21] IETF RFC 2805: "Media Gateway Control Protocol Architecture and Requirements".

[22] ITU-T Recommendation H.248.1: "Gateway control protocol".

[23] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[24] IETF RFC 3323: "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[25] IETF RFC 3325: "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

[26] ETSI TS 183 006: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services; Message Waiting Indication (MWI): Protocol specification".

[27] ETSI TS 183 011: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Anonymous Communication Rejection (ACR) and Communication Barring (CB); Protocol specification".

[28] ETSI ES 282 010: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Charging management [Endorsement of 3GPP TS 32.240 Release 7, 3GPP TS 32.260 Release 7, 3GPP TS 32.297 Release 7, 3GPP TS 32.298 Release 7 and 3GPP TS 32.299 Release 7, modified]".

[29] ITU-T Recommendation Q.763: "Signalling System No. 7 - ISDN User Part formats and codes".

[30] ITU-T Recommendation Q.764: "Signalling System No. 7 - ISDN User Part signalling procedures".

[31] ITU-T Recommendation Q.1980.1: "The Narrowband Signalling Syntax (NSS) - Syntax definition".

[32] ETSI ES 283 027: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified]".

[33] ETSI TS 183 023: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services; Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating NGN PSTN/ISDN Simulation Services".

[34] ETSI EN 300 356 (all parts): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface".

[35] ITU-T Recommendation Q.735.3: "Multi-level precedence and preemption".

[36] ITU-T Recommendation Q.735.6: "Global Virtual Network Service (GVNS)".

[37] ITU-T Recommendation Q.736.3: "Reverse charging (REV)".

[38] IETF RFC 3261: "SIP: Session Initiation Protocol".

[39] ETSI EN 301 798: "Services and Protocols for Advanced Networks (SPAN); Anonymous Call Rejection (ACR) Supplementary Service; Service description".

[40] ETSI TS 183 036 *V2.1.1 (2009-01)*: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification". *(1 TR 127: Annex A; Modified version for VGW (IAD) connected to accesses of Deutsche Telekom only !)*

[41] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".

[42] ETSI TS 183 058: "Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); SIP Transfer of IP Multimedia Service Tariff Information; Protocol specification".

[43] ETSI ES 283 035: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Network Attachment Sub-System (NASS); e2 interface based on the DIAMETER protocol".

[44] ETSI TS 124 615: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.615 version 8.0.1 Release 8)".

[45] IETF RFC 3023: "XML Media Types".

[Ref_dt1] DT 1 TR 114: Technical Specification of the SIP (Gm) interface between the User Equipment (UE) and the NGN platform of Deutsche Telekom.

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

[i.1] IETF RFC 4825: "The Extensible Markup Language (XML) Configuration Access Protocol (XCAP)".

[i.2] IEEE 1003.1-2004: "Standard for information technology - portable operating system interface (POSIX). Shell and utilities".

[i.3] IETF RFC 4488: "Suppression of Session Initiation Protocol (SIP) REFER Method Implicit Subscription".

[i.4] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 8.5.0 Release 8)".

[i.5] IETF RFC 3515: "The Session Initiation Protocol (SIP) REFER Method".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

access gateway: gateway device that interworks a significant number of analogue lines/ISDN accesses (directly or via an V5 Access Network) to a packet network and is located at the operator's premises. An access gateway can take the form of a Media Gateway (A-MGW) or a Voice over IP Gateway (A-VGW).

loose coupling: on-hook and flash-hook are analyzed in the ~~AGCF/~~VGW; much like a simulation endpoint would operate

Media GateWay (MGW): gateway device acting at the media/transport plane, providing the functions of an MGF as defined in ES 282 001 [1]. A MGW may additionally relay signalling traffic, in which case it also provides the functions of an SGF as defined in ES 282 001 [1].

NOTE: In the present document, Media Gateway refers both to Access Gateways and to Residential Gateways, to form an A-MGW, or an R-MGW, respectively.

Media Gateway Controller (MGC): See Recommendation H.248.1 [22].

residential gateway: gateway device that interworks a small number of analogue lines/ISDN accesses

NOTE: A residential gateway typically contains one or two analogue lines or ISDN basic accesses and is located at the customer premises. A residential gateway can take the form of a Media Gateway (R-MGW) or a Voice over IP Gateway (R-VGW).

tight coupling: on-hook and flash-hook are interpreted by the AS

Voice over IP GateWay (VGW): SIP-based gateway device that implements both a media gateway function and a media gateway controller function as defined in RFC 2805 [21] and supports the provision of voice based services to analogue lines/ISDN accesses

NOTE: A Voice over IP Gateways (VGW) whether acting as an Access Voice over IP Gateway (A-VGW) or as a Residential Voice of IP Gateway (R-VGW) plays the role of a PES Endpoint (i.e. acting as an IMS UE with regards to the P-CSCF).

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

3PTY	Three-Party Service
ACR	Automatic Communication Rejection
AGCF	Access Gateway Control Function
AOC	Advice Of Charge
AS	Application Server
B2BUA	Back-to-Back User Agent
BGCF	Breakout Gateway Control Function
CCBS	Call Completion on Busy Subscriber
CCNR	Call Completion on No Reply
CD	Call Deflection
CFB	Call Forwarding on Busy
CFNR	Call Forwarding on No Reply
CFU	Call Forwarding Unconditional
CGP	Charging Determination Point
CLF	Connectivity session Location and repository Function
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CN	Core Network
COLP	COnnected Line identification Presentation
COLR	COnnected Line identification Restriction
CONF	CONFerence
CPG	Call ProGress
CSCF	Call Session Control Function
CUG	Closed User Group
CW	Call Waiting
DT	*Deutsche Telekom*
ECT	Explicit Call Transfer
FM	Feature Manager
FQDN	Fully Qualified Domain Name
GPL	Generic Parameter List
GVNS	Global Virtual Network Service
HOLD	call HOLD
IBCF	Interconnection Border Control Function

I-CSCF	Interrogating CSCF
IM	IP Multimedia
I-MGCF	Incoming - MGCF
IMS	IP Multimedia core network Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
MCID	Malicious Call Identification
MEGACO	MEdia GAteway COntrol protocol
MGC	Media Gateway Controller
MGCF	Media Gateway Control Function
MGF	Media Gateway Function
MGW	Media GateWay
MLPP	Multi-Level Precedence and Pre-emption
MRF	Multimedia Resource Function
MRFC	Multimedia Resource Function Controller
MRFP	Multimedia Resource Function Processor
MWI	Message Waiting Indicator
NGN	Next Generation Network
NSS	Narrowband Signalling Syntax
O-MGCF	Outgoing - MGCF
P-CSCF	Proxy - CSCF
PES	PSTN Emulation Subsystem
PSTN	Public Switched Telephone Network
REV	REVerse Charging
RFC	Request For Comments
S-CSCF	Serving CSCF
SDP	Session Description Protocol
SGF	Signalling Gateway Function
SIP	Session Initiation Protocol
SOC	Switch Order Command
SUB	SUBaddressing
TAS	Terminal Alerting Signal
TP	Terminal Portability
UA	User Agent
UE	User Equipment
UPSF	User Profile Server Function
URI	Uniform Resource Identifier
UUS	User-to-User Signalling
VGW	Voice over IP GateWay
VMS	Voice Mail System
XCAP	XML Configuration Access Protocol
XML	eXtensible Markup Language

4 IMS-based PSTN Emulation Subsystem (PES) overview

4.1 General

General modifications (valid for the whole document):

- *VGW procedures shall be used,*
- *AGCF is out of scope of this document,*
- *H.248 is not used,*
- *Overlap Signalling shall not be used,*
- *NSS is out of scope of this document,*

- *Subscription for profile delivery shall be supported.*
- *Where mentioned within this document the combination of the MGC and MG shall be seen as equivalent to the VGW functionality.*
- *The 199 (Early Dialog Terminated) response code is mandatory to understand.*

The IMS-based PSTN/ISDN Emulation Subsystem (PES) supports the emulation of PSTN/ISDN services for ~~*analogue/*~~ISDN terminals connected to the TISPAN NGN, through residential gateways ~~*or access gateways*~~. The IMS-based PES functional architecture is defined in [3].

Emulating PSTN/ISDN services using the IMS-based PES architecture assumes that the logic of the service to be emulated resides in one or more application servers playing the role of a PES application server.

~~*Analogue/*~~ISDN terminals are connected to residential gateways ~~*or access gateways*~~ using standard ~~*analogue/*~~ISDN interfaces. The protocol running on interfaces between these gateways and the PES is ~~*either the gateway control protocol according to ITU-T Recommendation H.248.1 [22] (P1 reference point) or*~~ the session initiation protocol (SIP) according to RFC 3261 [38] (Gm reference point)~~*, depending on the type of gateway*~~:

- ~~*H.248-based voice over IP media gateway (MGW); or*~~
- SIP-based voice over IP gateway (VGW).

~~*Media gateways incorporate the Media Gateway Functional (MGF) entity identified in ES 282 001 [1] and are controlled by an Access Gateway Control Function (AGCF), at the P1 reference point. Media gateways supporting ISDN accesses shall also incorporate a Signalling Gateway Function (SGF) as defined in ES 282 001 [1].*~~

Further details on the architecture are available in TS 182 012 [3].

~~*Annex C illustrates the use of the PES for implementing usual PSTN services identified in EG 201 973-2 [11].*~~

~~*Annex F specifies the interworking of analogue terminals using overlap dialling with SIP overlap signalling.*~~

4.2 URI and address assignments

In case multiple subscribers are connected to the same gateway, there is no need to allocate a private user identity per subscriber. Whether a private user identity is allocated per gateway, group of subscribers or per subscriber is a matter for each operator to decide.

~~*The AGCF stores private user identities and public user identities in a local data base.*~~

4.3 ~~*AGCF and*~~ VGW session and registration processing model

4.3.1 General

Figure 1 illustrates the session processing model used by the ~~*AGCF and*~~ VGW functional entities. ~~*An AGCF is modelled as comprising H.248 Media Gateway Controller (MGC), Feature Manager (FM), and SIP UA functionality. An AGCF interfaces to a Media Gateway (MG) and also to the S-CSCF (via P1 and Mw reference points respectively).*~~

A functional modelling of the VGW contains an entity similar to H.248 Media Gateway Controller, a Feature Manager, a SIP UA, and MGW functionality. The VGW interfaces to the P-CSCF using the Gm reference point.

NOTE: The internal architecture of functional entities is not standardized. Any internal interfaces are hidden and not testable.

The SIP UA functionality provides the interface to the other components of the IMS-based architecture. It is involved in registration and session processing as well as in event subscription/notification procedures with application servers.

The MGC functionality enable the session processing functionality to interface with existing line signalling such as ~~*analogue signalling or*~~ DSS1.

Session and registration processing in the ~~AGCF or~~ VGW involves only ~~two halves~~: ~~H.248 based MGC processing and~~ SIP User Agent (UA) processing (see figure 1), which ~~MGC processing focuses on the interactions with the media gateway functions, while SIP UA processing~~ focuses on the interactions with the IMS components. The Feature Manager (FM) coordinates the ~~two~~ processing activities.

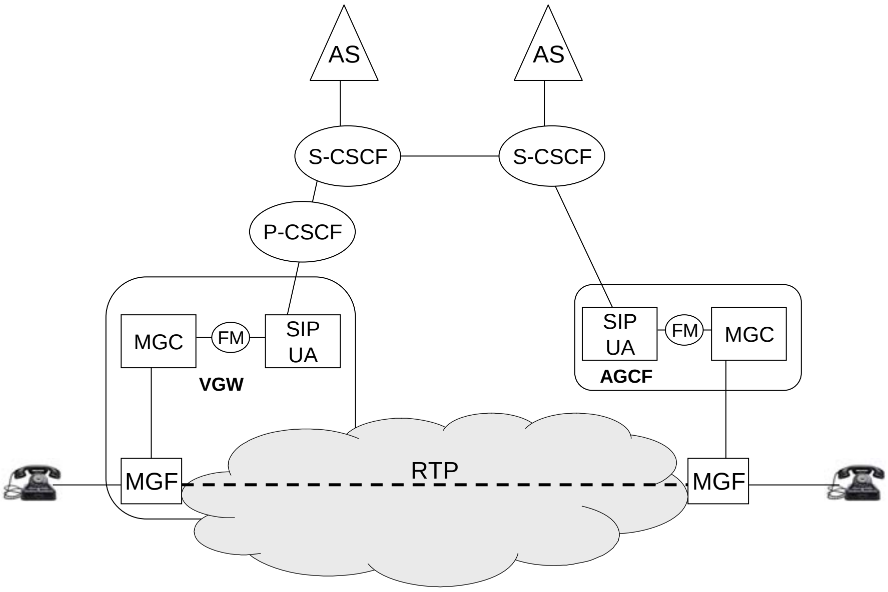

Figure 1: ~~AGCF/~~VGW session processing model

4.3.2 Conventions

The communication between the Feature Manager, MGC and SIP User Agent components of the ~~AGCF or~~ VGW is modelled using primitives. This modelling is used as an ease to describe the ~~AGCF and~~ VGW behaviour and is not intended to constrain implementations.

The following primitives are defined:

- Service-Change: This primitive is used by the MGC in the PES access point for reporting the ServiceChange event to the Feature Manager.
- Register-Request: This primitive is used by the Feature Manager for requesting the SIP User Agent to initiate appropriate SIP registration procedures on behalf of emulation users.
- Deregister-Request: This primitive is used by the Feature Manager for requesting the SIP User Agent to initiate appropriate SIP deregistration procedures on behalf of emulation users.
- Session-Attempt: This primitive is used to notify the Feature Manager of an outgoing call attempt.
- Setup-Request: This primitive is used to request establishment of a session.
- Setup-Response: This primitive is used to confirm the establishment of a session.
- Session-Update: This primitive is used to update a session description.
- Session-Progress: This primitive is used to report intermediate events during session establishment.
- Session-Release: This primitive is used to request the release of a session.
- Feature-Request: This primitive is used to report the occurrence of a service feature activation request from the user.

- Charging-Indication: This primitive is used to report the occurrence of a charging event.
- Service-Notification: This primitive is used to report the occurrence of a service notification.

The Feature Manager processes the above internal events received from the MGC side and request the SIP UA to generate the appropriate SIP messages based on the mapping described in table 1.

Table 1: Mapping from MGC side to SIP UA side

Internal primitive	SIP message
Setup Request	INVITE
Session Progress (alerting)	180 Ringing
Setup Response (no Answer)	480 (Temporary Not Available)
Setup Response (answer)	200 OK
Setup Response (busy)	486 Busy Here
Setup Response (reject)	606 Not Acceptable
Feature Request	re-INVITE
Release	BYE

The Feature Manager processes SIP messages received from the SIP UA side and transmit appropriate internal primitives to the MGC side, based on the mapping described in table 2.

~~*Exceptions to the above mapping applicable to calls originating from an analogue line are specified in the following clauses.*~~

Table 2: Mapping from SIP UA side to MGC side

SIP message	Internal primitive
INVITE	Setup Request
183 Session Progress	Session Progress
180 Ringing	Session Progress(alerting)
200 OK	Setup Response (answer)
603 (Decline), 408 (Request Timeout), 480 (Temporary Not Available)	Setup Response (no answer)
Other 4xx, 5xx, 6xx	Setup Response (failure)
486 (Busy Here) , 600 (Busy Everywhere)	Setup Response (busy)
re-INVITE with SDP, UPDATE	Session Update (SDP)
REFER	Session Update (refer)
INFO (charging) or NOTIFY (charging)	Charging Indication
NOTIFY (other)	Service Notification
BYE	Release

In case of ISDN access, the feature manager procedures shall conform to TS 183 036 [40].

5 Protocol using SIP and SIP events for PES

5.1 Introduction

This clause identifies the functional entities of the IMS-based PES architecture [3] that play a specific role in the implementation of PES services with regards to SIP processing.

5.2 Functional Entities

5.2.1 User Equipment (UE)

Conventional IMS UEs do not exist in PES. In PES, the User Equipment comprises one or more ~~*analogue/*~~ISDN terminals and may include the residential gateway to which they are connected. This residential gateway ~~*may be an H.248-controlled media gateway or*~~*is* a Voice over IP Gateway (VGW). Voice over IP Gateways (VGW) appear as conventional IMS UEs with regards to the P-CSCF, i.e. they play the role of a SIP user agent from SIP perspective. ~~Analogue/~~ISDN terminals are not visible to PES network entities.

~~NOTE: Analogue/ISDN terminals - "user equipment" - can also be connected to PES via an Access Gateway.~~

For the purpose of the PES, a residential VGW, in line with the behaviour of all VGWs, shall implement the role of a PES endpoint as described in clause 5.3.1.

Delete 5.2.2 Access Gateway Control Function (AGCF)

Delete 5.2.3 Application Server (AS)

Delete 5.2.4 Media Resource Function Controller (MRFC)

Delete 5.2.5 Media Gateway Controller Function (MGCF)

Delete 5.2.6 Interconnection border control function (IBCF)

5.2.7 Voice over IP gateway (VGW) acting as Access Gateway

For the purpose of the PES, an access VGW, in line with the behaviour of all VGWs, shall implement the role of the PES end point as described in clause 5.3.1.

The VGW entity encompasses the functionality of a Media Gateway Controller (MGC), Media Gateway (MG) and a SIP User Agent as defined in RFC 3261 [38]. Within the VGW, the MGC, MGW and SIP UA components are coordinated by a Feature Manager entity whose logical behaviour is described in annex B.

NOTE: The internal interfaces of the VGW are not standardized or testable.

5.3 Role

5.3.1 PES Endpoint

5.3.1.1 General

In addition to the procedures specified in the rest of clause 5.3.1, the PES endpoint shall support the procedures specified in ES 283 003 [4] and TS 183 028 [6] appropriate to an IMS UE.

5.3.1.2 Subscription for profile delivery

The PES Endpoint shall subscribe to the "ua-profile" event defined in [15] and support the Profile document defined in annex A of the present document and optionally to the "message-summary" event defined in RFC 3842 [18].

The subscription may be implicit or explicit. If explicit subscription is required, the identity of the AS acting as the profile delivery server where the subscription request shall be sent may be provisioned in the functional entity in which the PES endpoint is implemented. Alternatively, the user profile may contain an appropriate Initial Filter Criteria on SUBSCRIBE messages that ensure that such requests are sent to the AS acting as the profile delivery server.

For SIP UE supporting SIP-DSS1 interworking (e.g. IAD) the subscription of the "ua-profile" is implicit.

On receipt of a NOTIFY request reporting the "ua-profile" event and including an XML document with a Dial Tone Management element, the PES end point shall set the current dial tone to the value indicated by the dial-tone-pattern element received in the present document.

On receipt of a NOTIFY request reporting the "message-summary" event with a "Messages-Waiting" field set to "yes", the PES end point shall:

- Save the current dial tone value and set the current dial tone to the message waiting tone.
- ~~Send a Message Waiting Indicator message (see EN 200 659-3 [10]) to the calling user.~~

If the "message-summary" event is received with a "Messages-Waiting" field set to "no", the PES end point shall restore the previous current dial tone.

The contents of the SUBSCRIBE request shall be as follows:

- The value of the Request-URI shall be set to a provisioned or received value or the public user identity of the user to which the profile applies.
- The From and To header shall be set to the public user identity of the user to which the profile applies.
- The Accept header shall include the "application/simserv+xml".
- The Event header shall be set to the "ua-profile" event package.
- The Event parameters shall be set as follows:
    - The "profile-type" parameter shall be set to "user".

The "vendor", "model" and "version" parameter values shall be set to values specified by the implementer of the functional entity in which the PES endpoint is implemented, as specified in [15].

NOTE: When implicit subscription is used, the PES endpoint should also be prepared to accept "unsolicited" SIP NOTIFY requests with the "message-summary" event.

5.3.1.3 Registration procedures

The allocation of private and public user identities is left to each operator to decide. Two approaches are identified:

- ~~Group-based registration: One private user identity is assigned to a group of subscriber. A temporary public user identity is associated with this private user identity. Real public user identities representing the subscribers connected to the analogue or ISDN ports of the VGW are registered using implicit registration procedures defined in ES 283.003 [4].~~
- Line-based registration: A private user identity and one more public user identities are associated with each ~~analogue port/~~ISDN access connected to the VGW.

~~The two approaches are not mutually exclusive.~~

~~Depending on the registration method used, t~~The To and From headers in the REGISTER request shall be set to a SIP URI that contains the temporary public user identity associated with ~~the group to be registered or~~ the public user identity associated with the subscriber to be registered.

NOTE: The temporary public user identity may be provisioned on the VGW or ~~in case of line based registration~~ may be built from the line or access identifier (e.g. line-identifier@pes.operator.com).

Delete 5.3.1.4 Charging procedures

5.3.1.5 Outgoing Call

5.3.1.5.1 General

Calls initiated by a PES user via a PES end point shall operate following the UE call origination procedures defined in ES 283 003 [4].

Delete 5.3.1.5.2 Procedures for analogue lines

5.3.1.5.3 Procedures for ISDN access

The procedures specified TS 183 036 [40] apply.

If line-based registration is used, the PES Endpoint may set the P-Preferred-Identity header according to the calling party number received from the ISDN interface, as specified in TS 183 036 [40]. If the P-Preferred-Identity is not sent then the PES Endpoint shall ensure that the From header contains the equivalent identity. The PES Endpoint should verify that this identity is included in the P-Associated-URI header and otherwise overrides it with the first URI received in the P-Associated-URI.

If group-based registration is used the PES Endpoint should set the P-Preferred-Identity header to the default identity associated with the ISDN access where the call originates from. This default identity is derived from the ISDN access identifier through a provisioned mapping table or by creating a URI from the ISDN access identifier (e.g. isdn-access-id@pes.operator.com). In the later case, it is assumed that the PES AS will replace the contents of the P-Asserted-Identity with a meaningful public user identity (i.e. a Directory number).

NOTE: When the P-Preferred-Identity is set to an identity representing the ISDN access, the actual end user identity is still available in the From header as specified in TS 183 036 [40].

When performing interworking with the DSS.1 protocol, as specified in TS 183 036 [40], the PES end point shall indicate support of the application/vnd.etsi.pstn+xml MIME type.

Delete 5.3.1.5.4 Use of Overlap Signalling (Optional)

5.3.1.6 Terminating Call

5.3.1.6.1 General

Calls to a PES user via a PES end point shall operate following the UE call termination procedures defined in ES 283 003 [4].

Delete 5.3.1.6.2 Use of Overlap Signalling (Optional)

5.3.1.7 Supplementary services configuration

Delete 5.3.1.7.1 Analogue lines

5.3.1.7.2 ISDN lines

ISDN subscribers can configure supplementary services using procedures applied at either the Gm reference point or the Ut reference point.

In the case of the Ut reference point, HTTP PUT, HTTP GET or HTTP DELETE requests are used in accordance with RFC 4825 [i.1] and the supplementary services application usage specified in TS 183 023 [33].

In case of the Gm reference point two procedures are available;

- Use of service code commands as for analogue subscribers, when ISDN stimulus procedures are used by the end user terminal.
- Use of the MESSAGE method when ISDN functional procedures are used by the end user terminal.

The MESSAGE method is used to convey the service related information to the relevant target. The information to configure a service is embedded in an XML instance document contained in the MESSAGE request as specified in TS 183 036 [40].

The P-Preferred-Identity shall be set as specified in clause 5.3.1.5 for an INVITE request.

The To and From headers shall be set to the public user identity associated to the supplementary service action.

When receiving a MESSAGE request conveying a response to an ISDN supplementary service interrogation procedure (see TS 183 036 [40]), the PES endpoint shall look for the In-Reply-To header field and evaluate its contents to identify the initial MESSAGE request to which it is responding.

Delete Section 5.3.2 PES Access Point

Delete Section 5.3.3 PES Application Server

Delete Section 5.3.4 PES Announcement Server

6 Protocol using SIP/SDP for PES

6.1 Introduction

This clause identifies the functional entities of the IMS-based PES architecture [3] that play a specific role in the provision of PES services with regards to SDP processing in the context of SIP signalling.

6.2 Functional Entities

6.2.1 User Equipment (UE)

Conventional SIP UEs do not exist in PES (see clause 5.2.1).

For the purpose of the PES, the VGW shall implement the role of a PES endpoint as described in clause 6.3.1.

Delete Section 6.2.2 Access Gateway Control Function (AGCF)

Delete Section 6.2.3 Application Server (AS)

Delete Section 6.2.4 Media Resource Function Controller (MRFC)

6.2.5 Voice over IP gateway (VGW)

For the purpose of the PES, the VGW shall implement the role of the PES end point as described in clause 6.3.1.

The VGW entity encompasses the functionality of a Media Gateway Controller (MGC), Media Gateway (MG) and of SIP User Agent.

6.3 Roles

6.3.1 PES Endpoint

6.3.1.1 General

In addition to the procedures specified in the rest of clause 6.3.1, the PES endpoint shall support the procedures specified in ES 283 003 [4] appropriate to an IMS UE.

6.3.1.2 Originating Calls

When sending an SDP payload in a SIP message, the PES endpoint shall not include the "i=", "u=", "e=", "p=", "r=", and "z=" lines in the SDP, and it shall ignore them when received in the SDP.

~~*For calls originating from an analogue access, PES endpoint shall build an SDP offer as follows:*~~

- ~~*Only one media description shall be included (i.e. one m= line).*~~
- ~~*The media description shall contain the audio codecs supported and the MIME subtype "telephone-event" as described in RFC 4733 [17] , unless ITU-T Recommendation G.711 [23] is the only proposed codec.*~~

~~*NOTE: Support of T.38 is an option. outside the scope of TISPAN NGN present Release.*~~

For calls originating from an ISDN access, the PES endpoint shall build an SDP offer according to TS 183 036 [40].

6.3.1.3 Terminating Calls

6.3.1.3.1 General

When sending an SDP payload in a SIP message, the PES endpoint shall not include the "i=", "u=", "e=", "p=", "r=", and "z=" lines in the SDP, and it shall ignore them when received in the SDP.

When the PES endpoint sends a 183 (Session Progress) response with SDP payload, it shall only request confirmation for the result of the resource reservation at the originating endpoint if there are any remaining unfulfilled preconditions.

Delete Section 6.3.1.3.2 Analogue Access

6.3.1.3.3 ISDN Access

Processing of media description for calls terminating on an ISDN access shall conform to TS 183 036 [40].

Delete Section 6.3.2 PES Access Point

Delete Section 6.3.3 PES Application Server

Delete Section 6.3.4 PES Announcement Server

Delete Section 7 Protocol using H.248 for PES

8 Protocol using DSS1 for PES

8.1 Introduction

This clause identifies the functional entities of the IMS-based PES architecture [3] that play a specific role in the provision of PES services with regards to DSS.1 processing.

8.2 Functional Entities

8.2.1 User Equipment (UE)

Conventional SIP UEs do not exist in PES (see clause 5.2.1).

For the purpose of the PES, the VGW shall implement the role of a PES endpoint as described in clause 8.3.1.

Delete Section 8.2.2 Access Gateway Control Function (AGCF)

Delete Section 8.2.3 Signalling Gateway Function (SGF)

8.2.4 Voice over IP gateway (VGW)

For the purpose of the PES, the VGW shall implement the role of the PES endpoint as described in clause 8.3.1.

The VGW entity encompasses the functionality of a Media Gateway Controller (MGC), Media Gateway (MG) and a SIP User Agent.

8.3 Roles

8.3.1 PES Endpoint

The PES Endpoint shall support the DSS1 protocol according to EN 300 403-1 [41].

8.3.2 PES Access Point

The PES Access point shall support the DSS1 protocol according to EN 300 403-1 [41] and one or more backhaul procedures described in ES 283 002 [5] for relaying ISDN D-channel information, depending on the type of ISDN access supported.

8.3.3 PES Signalling Gateway

The PES signalling gateway shall support one or more backhaul procedures described in ES 283 002 [5] for relaying ISDN D-channel information, depending on the type of ISDN access supported and one or more backhaul procedures described in ES 283 002 [5] for relaying ISDN D-channel information, depending on the type of ISDN access supported.

Annex A (normative, mandatory): XML document structure for Profile Delivery

Profile documents are sub-trees of the *simservs* XML document defined in TS 183 023 [33]. The following schema shall be used to describe XML documents that specify the profile elements applicable to an endpoint.

```
<?xml version="1.0" encoding="UTF-8"?>
<xs:schema xmlns:ss="http://uri.etsi.org/ngn/params/xml/simservs/xcap"
xmlns:xs="http://www.w3.org/2001/XMLSchema"
targetNamespace="http://uri.etsi.org/ngn/params/xml/simservs/xcap" elementFormDefault="qualified"
attributeFormDefault="unqualified">
   <xs:element name="dial-tone-management" substitutionGroup="ss:absService">
       <xs:annotation>
           <xs:documentation>Dial Tone Management
            </xs:documentation>
       </xs:annotation>
       <xs:complexType>
           <xs:complexContent>
               <xs:extension base="ss:simservType">
                   <xs:sequence>
                       <xs:element name="dial-tone-pattern" default="standard-dial-tone"
minOccurs="0">
                           <xs:simpleType>
                               <xs:restriction base="xs:string">
                                   <xs:enumeration value="standard-dial-tone"/>
                                   <xs:enumeration value="special-condition-tone"/>
                                   <xs:enumeration value="message-waiting-tone"/>
                               </xs:restriction>
                           </xs:simpleType>
                       </xs:element>

   <xs:element name="mcid-service" default="mcid-service-withdrawn" minOccurs="0">
       <xs:simpleType>
           <xs:restriction base="xs:string">
               <xs:enumeration value="mcid-service-provisioned"/>
               <xs:enumeration value="mcid-service-withdrawn"/>
           </xs:restriction>
       </xs:simpleType>
                        </xs:element>

   <xs:element name="no-dialling-behaviour" default="rejectCall" minOccurs="0">
       <xs:simpleType>
           <xs:restriction base="xs:string">
               <xs:enumeration value="rejectCall"/>
               <xs:enumeration value="immediateCallSetup"/>               <xs:enumeration
value="deferredCallSetup"/>
           </xs:restriction>
       </xs:simpleType>
   </xs:element>

   <xs:element name="hold-service" default="hold-service-provisioned" minOccurs="0">
       <xs:simpleType>
           <xs:restriction base="xs:string">
               <xs:enumeration value="hold-service-provisioned"/>
               <xs:enumeration value="hold-service-withdrawn"/>
           </xs:restriction>
       </xs:simpleType>
   </xs:element>
   <xs:element name="toggle-service" default="toggle-service-withdrawn" minOccurs="0">
       <xs:simpleType>
           <xs:restriction base="xs:string">
               <xs:enumeration value="toggle-service-provisioned"/>
               <xs:enumeration value="toggle-service-withdrawn"/>
           </xs:restriction>
       </xs:simpleType>
   </xs:element>
   <xs:element name="three-pty-service" default="three-pty-service-withdrawn" minOccurs="0">
       <xs:simpleType>
           <xs:restriction base="xs:string">
               <xs:enumeration value="three-pty-service-provisioned"/>
               <xs:enumeration value="three-pty-service-withdrawn"/>
           </xs:restriction>
       </xs:simpleType>
```

```
~~    </xs:element>~~
~~    <xs:element name="cw-service" default="cw-service-provisioned" minOccurs="0">~~
~~        <xs:simpleType>~~
~~            <xs:restriction base="xs:string">~~
~~                <xs:enumeration value="cw-service-provisioned"/>~~
~~                <xs:enumeration value="cw-service-withdrawn"/>~~
~~            </xs:restriction>~~
~~        </xs:simpleType>~~
~~    </xs:element>~~
~~    <xs:element name="priority-line" default="priority-line-disabled"  minOccurs="0">~~
~~        <xs:simpleType>~~
~~            <xs:restriction base="xs:string">~~
~~                <xs:enumeration value="priority-line-enabled"/>~~
~~                <xs:enumeration value="priority-line-disabled"/>~~
~~            </xs:restriction>~~
~~        </xs:simpleType>~~
~~     </xs:element>~~
                   </xs:sequence>
               </xs:extension>
           </xs:complexContent>
       </xs:complexType>
   </xs:element>
</xs:schema>
```

Annex B (normative): ~~AGCF/~~VGW Feature Manager

B.1 Void

B.2 Void

B.3 Void

B.4 Feature manager behaviour

B.4.1 Registration procedures

Based on the information received from the line side signalling and local configuration data (mapping between line identities and IMS identities, authentication parameter, etc.) the Feature Manager requests the SIP UA component to initiate appropriate SIP registration procedures (per line registration, group registration).

Delete Section B.4.1.1 Group registration procedures

B.4.1.2 Per line/access registration procedures

There may a lot of terminations to register and deregister at the same time from the MGC side. In order to avoid the impact of the flood registration and deregistration on the IMS core network, the Feature Manager may use some mechanism to control the number of registration/deregistration during a specified period of time.

B.4.1.2.1 User-initiated registration

On receipt of a **Service-Change** primitive from the MGC component with ServiceChangeMethod parameter set to "Restart" indicating that service will be restored on the specified Terminations, the Feature Manager shall:

- Lookup the configured data for the public user identities and the private user identities of the related users.
- If the related users have not been registered, use a Register-Request primitive to request the SIP UA to initiate appropriate SIP registration procedures.

B.4.1.2.2 User-initiated deregistration

On receipt of a Service-Change primitive from the MGC component with ServiceChangeMethod parameter set to "Graceful" or "Forced" indicating that the specified Terminations will be taken out of service, the Feature Manager shall:

- Lookup the configured data for the public user identities and the private user identities of the related users.
- If the related users have been registered, use a Deregister-Request primitive to request SIP UA to initiate appropriate SIP deregistration procedures.

Delete Section B.4.1.2.3 Exception procedures

Delete Section B.4.2 Flash Hook Management

Delete Section Annex C (informative): Implementation of Supplementary Services

Delete Section Annex D (normative): Mapping between SIP and the subscriber line protocol

Delete Section Annex E (normative): AOC - Extended XML schema (version 2)

Mapping of AOC please see 1TR127 ANNEX A

Annex F (normative): Overlap Sending

~~*This annex describes the handling of Overlap Sending according to EN 300 403-1 [41], clause 5.1.3.*~~

Within the context of this specification this annex describes the convertion of overlap dialling to en-bloc signalling.

F.0 General

Three methods of signalling are described in this annex:

- F.1 describes sending of Invite with determining the end of address signalling.
- ~~*F.2 describes a signalling procedure without determining the end of address signalling using a multiple INVITE method.*~~
- ~~*F.3 describes a signalling procedure without determining the end of address signalling using a In-Dialog method.*~~

The collection of Digits at the originating VGW~~*/AGCF*~~ shall be controlled using the following timers:

- T-FirstDigit timer: The amount of time allowed for the user to enter the first digit.
- T*a1*~~*-InterDigit timer*~~: The amount of time allowed for the user to enter each subsequent digit.
- Additional timers which are per signalling method are described in the relevant clauses F.1~~*, F.2 and F.3*~~.

~~*The Ta3 timer used in clauses F.2 and F.3, shall be provisioned to a value that is greater than the T-InterDigit timer.*~~

Clause F.4 provides a table containing the complete list of Timers.

F.1 Sending of Invite with determining the end of address signalling

F.1.1 Actions at the originating VGW~~/AGCF~~

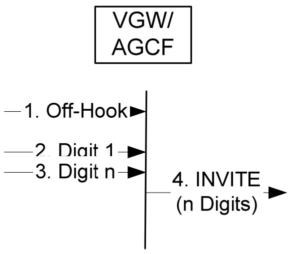

Figure F.1: Receipt of Digit information at the originating VGW~~/AGCF~~

After initiating the normal incoming PSTN call establishment procedures, determining the end of address signalling and selecting to route the call to the IMS domain, the originating VGW~~*/AGCF*~~ sends the initial INVITE. The initial INVITE contains all digits, i.e. en-bloc sending.

The end of address signalling is determined by the earlier of the following criteria:

~~*a) by receipt of the maximum number of digits used in the national numbering plan; or*~~

b) by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party. This could be achieved by analysis of a provisioned dial plan; or

Note: This applies only in cases of emergency calls (110 and 112) where Digit map is used.

c) by observing that timer Ta1 has expired after the receipt of the latest received address digit and the minimum number of digits required for routing the call have been received~~.~~,
or

Note: The default value for the minimum number of Digits is 1.

d) Receipt of the End of Dialling Digit “#”.

NOTE 1: When “#” is part of a Service Code, it shall not be interpreted as End of Dialling Digit. It shall be sent as part of the Request SIP URI and the To header Field.

NOTE 2: When “#” is interpreted as End of Dialling Digit, it shall not be sent as part of the Request SIP URI and the To header Field.

~~*If the end of the address signalling is determined in accordance with criteria a) or b), the timer Ta2 is started when INVITE is sent.*~~

The use of the Digit map mechanisms for emergency calls (only dialled numbers 110 and 112) is strongly recommended.

The ~~*AGCF/*~~VGW can contain a configurable digit map which is used to analyse the received address digits. This digit map can be used to identify the required number of digits to be entered for a particular digit sequence. ~~*The procedures for Digit maps are described within clause 7.3.1.3.1.1.*~~

Even in the absence of a digit map, it is appropriate for the ~~*AGCF/*~~VGW to collect dialled digits. The ~~*AGCF/*~~VGW shall contain a configurable parameter indicating the minimum number of digits expected in the sequence of Called party number information elements before a Request-URI is constructed and an INVITE request sent. The minimum number could be zero.

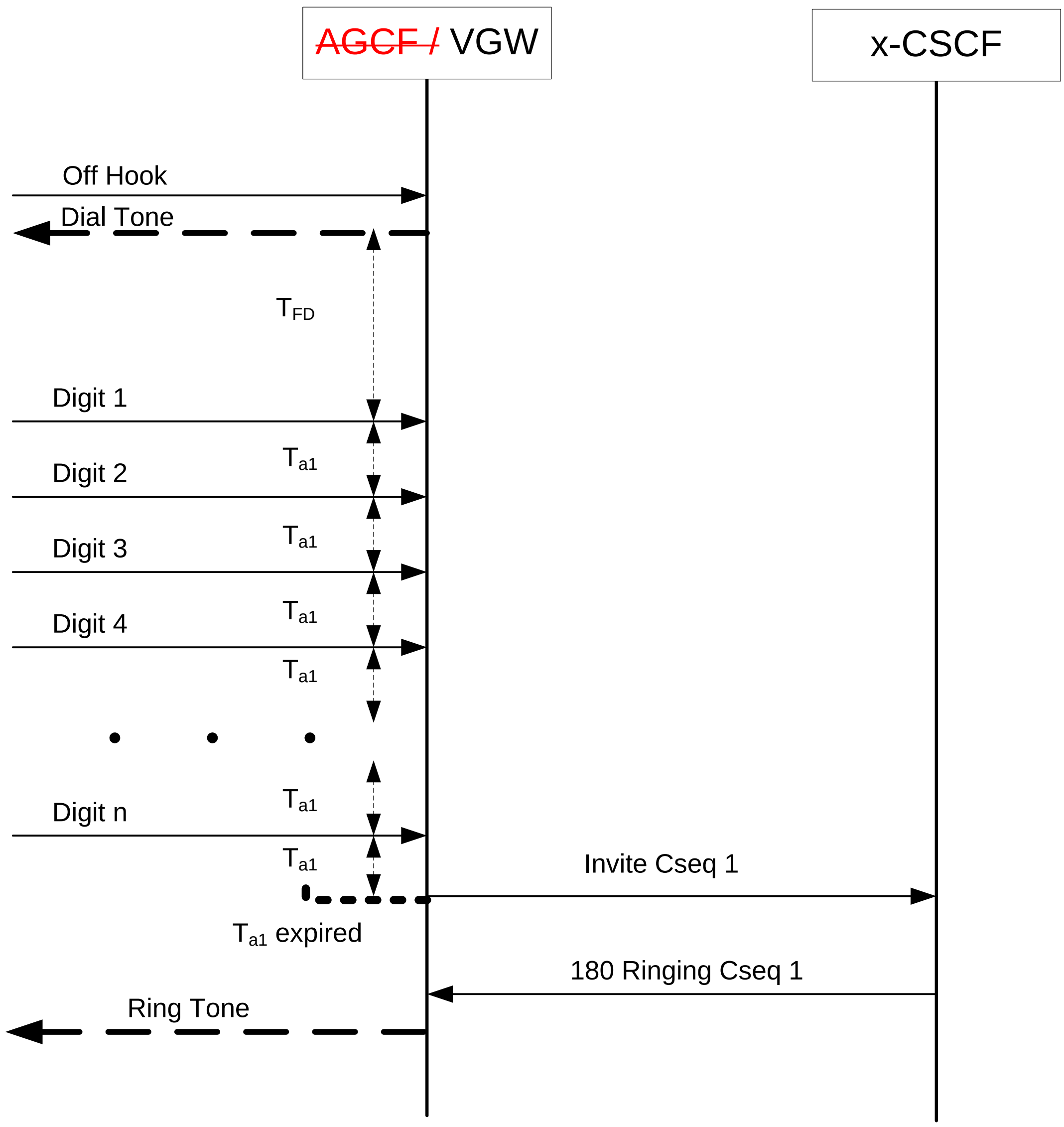

Figure F.1-2: Overlap with Ta1

F.1.2 Actions at the terminating VGW~~/AGCF~~

No action with regard to overlap is needed.

DDI is out of scope of the present document.

Delete Section F.2 multiple INVITE Overlap Dialling Procedures (Optional)

Delete Section F.3 In-Dialog Method (Optional)

F.4 Timers

Table F.1: Timers for interworking

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry
T-FirstDigit	1 s to 99 s (default=60 s)	Upon start of dial tone injection	On receipt of the first dialed digit	Release Call
~~T-InterDigit~~	~~1 s to 15 s (default=10 s)~~	~~Upon receipt of a new address message~~	~~Upon receipt of subsequent address message or 180 Ringing or 183 Session Progress with P-Early Media header authorizing early media~~	~~a) Disable Digit Receiver; b) if Ta3 is not running then Release Call~~
Ta1	4 s to ~~6~~ 12 s (default of 4 s) (see note 1)	At the receipt of fresh address information. ~~When last address information is received and the minimum number of digits required for routing the call have been received.~~	At the receipt of fresh address information.	a) Disable Digit Receiver b) Send INVITE
~~Ta2 (Multiple Invite)~~	~~10 s to 15 s (see notes 2 and 4)~~	~~When INVITE is sent.~~	~~On reception of 180 Ringing , or 183 Session Progress, or 404 Not Found or 484 Address Incomplete for an INVITE transaction for which Ta2is running, or 200 OK (INVITE).~~	~~Release Call~~
~~Ta2 (In-Dialog)~~	~~10 s to 15 s (see notes 2 and 4)~~	~~When INVITE is sent or when INFO is sent.~~	~~On reception of 180 Ringing , or 183 Session Progress, or 404 Not Found or 484 Address Incomplete for an INVITE transaction for which Ta2 is running, or 200 OK (INVITE) or 200 OK (INFO).~~	~~Release Call~~
~~Ta3 (Multiple invite)~~	~~4 s to 20 s (default of 15 s) (see note 3)~~	~~On receipt of 404 Not Found or 484 Address Incomplete if there are no other pending INVITE transactions for the corresponding call.~~	~~At the receipt of fresh address information~~	~~Release call~~
~~Ta3 (In-Dialog)~~	~~4 s to20 s (default of 15 s) (see note 3)~~	~~On receipt of 404 Not Found or 484 Address Incomplete if there are no other pending INVITE transactions for the corresponding call. On receipt of 200 OK (INFO), or 183 Session Progress without early media authorization.~~	~~At the receipt of fresh address information or 180 Ringing or 183 with P-Early-Media header authorizing early media or 200 OK (INVITE)~~	~~Release call~~

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry
~~Ta4~~	~~0,5 s to -4 s~~	~~On receipt of the initial address information..~~	~~At expiry~~	~~Send INVITE~~

NOTE 1: This timer is used in clause F.1 when overlap signalling is received from access line and converted to en-block signalling at the ~~AGCF/~~VGW.

~~NOTE 2: This timer is used in clauses F.2 and F.3 to wait for an 404/484 response. In addition clause F.3 uses this timer for:~~

~~a) to wait for an 183 response to an INVITE;~~

~~b) to wait for a 200 OK (INFO) response.~~

~~NOTE 3: This timer is known as the "SIP dialog protection timer". This timer is only used where the AGCF/VGW is configured to send INVITE before end of address signalling is determined. The Ta3 timer shall be greater than the T-InterDigit timer.~~

~~NOTE 4: The value of timer Ta2 may vary beyond these limits, e.g. as a result of called party number analysis.~~

Delete Section Annex G (informative): Digit collection in MRF after receipt of flash-hook in the tight coupling model

Annex H (informative): Bibliography

- ETSI ES 282 007: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Subsystem (IMS); Functional architecture".

Annex I (informative): Change history

Date	WG Doc.	CR	Rev	CAT	Title / Comment	Current Version	New Version
01-05-06					Publication of v1.1.1		
07-03-08	16bTD512	003		F	Alignment with Stage 2	2.0.0	2.1.0
07-03-08	WG3-02-016r 2 in 16bTD264	004		F	Support of RTTI	2.0.0	2.1.0
07-03-08	WG3-02-017r 2 in 16bTD264	005		F	Dial tone management	2.0.0	2.1.0
07-03-08	WG3-02-034r 2 in 16bTD264	006		B	P-Access-Network-Info	2.0.0	2.1.0
07-03-08	16bTD301r4	007	1	F	AOC for POTS Line Metering	2.0.0	2.1.0
07-03-08		008			Withdrawn because replaced by CR 007r1	2.0.0	2.1.0
07-03-08	WG3-02-041r 2 in 16bTD264	009		F	P-Earlymedia header	2.0.0	2.1.0
07-03-08	16bTD267r3	010	3	F	Population of P-Asserted-Id	2.0.0	2.1.0
07-03-08	16bTD268r1	011		F	Switching order commands examples	2.0.0	2.1.0
07-03-08	16bTD269r1	012		F	Actions performed by AGCF	2.0.0	2.1.0
07-03-08	16bTD275r3	013	3	C	Call Initiation	2.0.0	2.1.0
07-03-08	16bTD270r2	014		F	Handling of suspend timer	2.0.0	2.1.0
30-05-08	17bTD197r2	015		B	Profile delivery	2.1.0	2.1.1
30-05-08	17bTD218r2	016		F	Pulse Metering - new section C.16 in TS 183 043	2.1.0	2.1.1
30-05-08	17bTD234r1	018		D	Call Waiting	2.1.0	2.1.1
30-05-08	17bTD235r1	019		D	MCID	2.1.0	2.1.1
30-05-08	17bTD236r1	020		F	CLIP CLIR	2.1.0	2.1.1
30-05-08	17bTD238r1	021		D	WI03134 Call Forwarding	2.1.0	2.1.1
30-05-08	17bTD239r1	022		F	WI03134 Automatic Call Return	2.1.0	2.1.1
30-05-08	17bTD240r1	023		F	ACR	2.1.0	2.1.1
30-05-08	17bTD241r1	024		F	AGCF Charging	2.1.0	2.1.1
30-05-08	17bTD242r1	025		F	Session Progress	2.1.0	2.1.1
30-05-08	17bTD257r2	026		F	3PTY	2.1.0	2.1.1
17-06-08					Clean-up and addition of Change History annex by ETSI Secretariat	2.1.1	2.1.2
02-07-08	18WTD202r1 WG3-03-021r 4	027	1	F	AGCF behaviour when the user does not dial any digit	2.1.2	2.1.3
02-07-08	18WTD132 WG3-03-022r 1	028		F	Separate the loose and tight coupled options in the 3PTY service	2.1.2	2.1.3
02-07-08	18WTD132 WG-03-023r1	029		F	Separate the loose and tight coupling for CW	2.1.2	2.1.3
02-07-08	WG3-03-016r 3	030		F	Advice of Charge (Annex C.2)	2.1.2	2.1.3
02-07-08	WG3-03-017r 3 revised in xxx	031	1	F	AOC-Extended XML schema	2.1.2	2.1.3
02-07-08	18WTD132 WG3-03-024r 1	032		C	Additional flexibility of the AGCF to support different message servers	2.1.2	2.1.3
02-07-08	18WTD132 WG3-03-027r 1	033		F	Correction to the use of privacy field in ACR	2.1.2	2.1.3
02-07-08	18WTD122r1	034	1	C	Overlap in-dialog method	2.1.2	2.1.3
02-07-08	18WTD202r1 WG3-03-033r 1	035		D	Harmonization of global and group registration	2.1.2	2.1.3
02-07-08	18WTD202r1 WG3-03-034r 1	036		F	Addition of figures for 3PTY	2.1.2	2.1.3
02-07-08	18WTD202r1 WG3-03-035r 1	037		B	Wildcarded Identities	2.1.2	2.1.3
02-07-08	18WTD123r3	038		C	Additional procedures to align the loose coupled procedures with the MCID simulation service	2.1.2	2.1.3
02-07-08	18WTD192r1	039		D	Profile Delivery Editorial Corrections	2.1.2	2.1.3
02-07-08	18WTD193r2	040		B	Profile Item HOLD	2.1.2	2.1.3
02-07-08	18WTD194r2	041		B	Profile Item Toggle	2.1.2	2.1.3
02-07-08	18WTD195r2	042		B	Profile Item 3PTY	2.1.2	2.1.3
02-07-08	18WTD196r2	043		B	Profile Item CW	2.1.2	2.1.3
02-07-08	18WTD253r1	044		B	Profile Delivery Optional	2.1.2	2.1.3
02-07-08	18WTD200r2	045		F	CLIP / CLIR section C.5	2.1.2	2.1.3
02-07-08	18WTD218r1	046		F	Multiple terminations per public user identity	2.1.2	2.1.3
02-07-08	18WTD246r1	047		F	Alignment between Clause 7 and Annex C	2.1.2	2.1.3
02-07-08	18WTD248r1	048		F	Alignment between Annex B and C	2.1.2	2.1.3
26-09-08	18bTD267r1	049		C	Overlap in-dialog method	2.1.3	2.1.4
26-09-08	18bTD262r1	050		F	Improving the structure of TS 183 043	2.1.3	2.1.4
26-09-08	18bTD254r3	051		B	Profile Delivery XML	2.1.3	2.1.4
26-09-08	18bTD278r3	053		C	VGW Procedures for tightly coupled and loosely coupled procedures	2.1.3	2.1.4
26-09-08	18bTD280r2	052		C	Call Waiting procedures	2.1.3	2.1.4

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05-11-08	19WTD202r1	054		F	Correction of the Name of P-Asserted_identity Header	2.1.3	2.1.5
05-11-08	19WTD129r1	055		F	P-Preferred-Identity and P-Asserted-Identity headers	2.1.3	2.1.5
05-11-08	19WTD138r1	056		F	VGW protocol role	2.1.3	2.1.5
05-11-08	19WTD180r1	057		F	AS behaviour	2.1.3	2.1.5
05-11-08	19WTD107r2	058	2	F	Corrections to VGW	2.1.3	2.1.5
05-11-08	19WTD124r2	059		F	Annex A - Profile Delivery XML	2.1.3	2.1.5
05-11-08	19WTD200r2	060	1	F	Flash Hook procedures	2.1.3	2.1.5
05-11-08	19WTD127r2	061		F	CLIP/CLIR -section C.5.3	2.1.3	2.1.5
05-11-08	19WTD131r1	062		F	SS Configuration	2.1.3	2.1.5
05-11-08	19WTD166r2	063		F	HOLD procedures in Tight coupling mode	2.1.3	2.1.5
05-11-08	19WTD197r2	064	1	F	Call waiting Procedures	2.1.3	2.1.5
05-11-08	19WTD201r1	065		F	3PTY/CONF Procedures	2.1.3	2.1.5
05-11-08	19WTD139r2	066		F	SDP Procedures	2.1.3	2.1.5
05-11-08	19WTD212r1	067		F	OVERLAP AS procedure	2.1.3	2.1.5
05-11-08	19WTD126r1	068		F	Overlap Signalling Ta3 timer	2.1.3	2.1.5
05-11-08	19WTD128r1	069		F	Overlap sending	2.1.3	2.1.5
05-11-08	19WTD132r1	070		F	Correction of Call Flow for In-Dialog Method, INVITE without digits at off-hook	2.1.3	2.1.5
05-11-08	19WTD206r4	071	2	F	Correction of Call Flow for In-Dialog Method, INVITE without digits at off-hook	2.1.3	2.1.5
05-11-08	19WTD160r1	072			Backward Compatibility of Tight Coupling Procedures	2.1.3	2.1.5
05-11-08	19WTD125r2	073	1	F	Annex E - Extended AOC XML schema naming	2.1.3	2.1.5
05-11-08	19WTD170r3	074		F	General Explanation of loose and tight coupling model - NOT IMPLEMENTED in 2.1.5	2.1.3	2.1.5
05-11-08	19WTD163r4	075		F	Flash hook handling in Tight coupling model	2.1.3	2.1.5
13-11-08					CRs 049 to 075 TB approved and clean-up by ETSI Secretariat	2.1.5	2.2.0
24-11-08	19bTD210	054	1	F	Correction of the Name of P-Asserted_identity Header	2.2.0	2.2.1
24-22-08					Correction implementation of CR055	2.2.0	2.2.1
05-11-08	19WTD170r3	074		F	General Explanation of loose and tight coupling model - Already TB approved at TISPAN#19 but not implemented	2.2.1	2.2.2
26-11-08	19bTD104r2	076		F	MESSAGE correlation	2.2.1	2.2.2
26-11-08	19bTD151r1	077		F	Changes on timer section	2.2.1	2.2.2
26-11-08	19bTD095r2	078	1	B	Re-ringing for loose coupled	2.2.1	2.2.2
26-11-08	19bTD110r3	080	2	F	Registration of MIME type with IANA	2.2.1	2.2.2
26-11-08	19bTD216r5	081	4	F	Call Waiting procedures	2.2.1	2.2.2
26-11-08	19bTD170r1	082		F	Correction to In Dialog Method	2.2.1	2.2.2
26-11-08	19bTD150r1	083		F	Correction of Call Flow for In-Dialog Method - sequence numbering	2.2.1	2.2.2
26-11-08	19bTD149r1	084		B	Message body to transfer digits In-Dialog	2.2.1	2.2.2
26-11-08	19bTD232r2	085		F	Clarification and editorial changes to WI3134 V2.2.1	2.2.1	2.2.2
09-12-08	19bTD241r1	086		F	Re-ringing	2.2.2	2.2.3
					Publication	2.2.3	2.3.1
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History

Document history		
V1.1.1	May 2006	Publication
V2.3.1	March 2009	Publication
V1.0.0	**June 2010**	**First edition of 1 TR 127: Annex B;** Modified version for VGW (IAD) connected to accesses of Deutsche Telekom only ! - All modifications to the present document (endorsement) defined in the former version of 1TR127 are taken out and incorporated in the present document; - Additional requirements and information incorporated (in line with 1TR126 Annex A); - Annex C deletet (Figure C.6 copied to Annex A); - Annex D deleted; - Annex F and F.1.1 partly modified; - Figure F.1-2 inserted; - Formal changes; - Document properties updated; - Contents corrected; - History updated; - T-Home replaced by Deutsche Telekom.
V1.1.0	**December 2011**	Editorial change, deletion of all stricken text to delete redundant text that does not apply.