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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

RTS/TISPAN-03134-NGN-R2

Keywords

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Contents

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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 FMED15 of Deutsche Telekom Netzproduktion GmbH, Fixed Mobile Engineering Deutschland (in the following named as Deutsche Telekom) and defines the emulation services for IP Multimedia Subsystem (IMS) provided by a SIP User Equipment (UE) via an analogue (POTS) interface (e.g. Integrated Access Device: IAD)

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~~).

Where whole sections do not apply the section is marked as deleted instead of showing the whole deleted section

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:
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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: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".

[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".

[46] ETSI TS 124 447 V8.0.0 (2008-04): Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; NGN IMS Supplementary Services; Advice Of Charge (AOC) (3GPP TS 24.447 version 8.0.0 Release 8)

[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 *en-block*~~*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 involve~~s two halves~~: ~~H.248 based MGC processing and~~ SIP User Agent (UA) processing (see figure 1). 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.

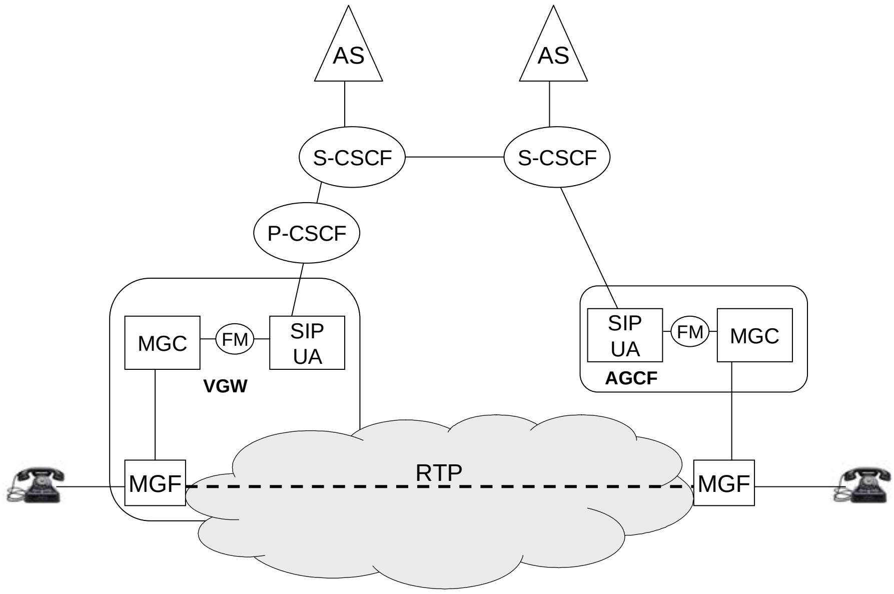

Figure 1: ~~AGCF/~~VGW session processing model

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 Section 5.2.2 Access Gateway Control Function (AGCF)

Delete Section 5.2.3 Application Server (AS)

Delete Section 5.2.4 Media Resource Function Controller (MRFC)

Delete Section 5.2.5 Media Gateway Controller Function (MGCF)

Delete Section 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-Analogue interworking (e.g. IAD) the subscription of the "ua-profile" is explicit.

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.

NOTE: *The Message Waiting Indicator message (according to EN 200 659-3 [10]) is currently not used within Deutsche Telekom.*

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, 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 Section 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].

5.3.1.5.2 Procedures for analogue lines

On receipt of a non-empty Setup-Request the PES Endpoint shall send an INVITE request with the call destination in the Request-URI, an SDP Offer for a voice call. The INVITE may include a P-Preferred-Identity header field set to the default identity associated with the analogue termination where the off-hook event was detected. If the P-Preferred-Identity is not sent then the PES Endpoint shall ensure that the From header contains the equivalent identity.

If line-based registration has been used this default identity is the first one received in the P-Associated-URI header.

SIP UE supporting SIP-Analogue interworking (e.g. IAD) shall have the possibility to process a received P-Associated-URI header.

~~If group-based registration has been used, this default identity is derived from the analogue termination identifier through a provisioned mapping table or by creating a URI from the analogue termination identifier (e.g. termination-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).~~

~~On receipt of an empty Setup-Request (i.e. without explicit destination information) the PES Endpoint shall build and send an INVITE Request as described above except that the Request-URI shall be set to a provisioned value (e.g. nodial@pes.operator.com). It is up to the PES Application Server to reject the call by sending back an appropriate response code or to substitute the Request-URI with a subscriber-specific default value before forwarding the INVITE requests towards a destination.~~

Delete Section 5.3.1.5.3 Procedures for ISDN access

Delete Section 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 Section 5.3.1.6.2 Use of Overlap Signalling (Optional)

5.3.1.7 Supplementary services configuration

5.3.1.7.1 Analogue lines

Analogue subscribers can control their supplementary services using service code commands as defined in ETS 300 738 [12] *and specified in Annex D of 1 TR 114 [Ref_dt1] for Deutsche Telekom* included in the user part of the Request-URI of an INVITE message. Further details are provided in annex C.

Delete Section 5.3.1.7.2 ISDN lines

Delete Section 5.3.2 PES Access Point

Delete Section 5.3.3 PES Application Server

Delete Section 5.3.4 PES Announcement Server

Delete Section 5.3.5 PES Interworking Application

Delete Section 5.3.6 PES interconnection application

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.

6.3.1.3.2 Analogue Access

For calls terminating on an analogue access, the PES endpoint shall:

- Reject any media descriptions whose media type is different from audio, by setting the corresponding port to 0.

NOTE: Support of T.38 is outside the scope of TISPAN NGN present Release.

- check for codecs that match the requested SDP, which may include the MIME subtype "telephone-event" as described in RFC 4733 [17].

When the PES endpoint generates and sends a 183 (Session Progress) response to an initial INVITE request, the PES endpoint shall:

- set SDP indicating the selected codec and the MIME subtype "telephone-event" as described in RFC 4733 [17] if received in the SDP offer.

Delete Section 6.3.1.3.3 ISDN Access

Delete Section 6.3.2 PES Access Point

Delete Section 6.3.3 PES Application Server

Delete Section 7 Protocol using H.248 for PES

Delete Section 8 Protocol using DSS1 for PES

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

B.4.2 Flash Hook Management

B.4.2.1 Void

B.4.2.2 Flash-Hook Management for analogue access

B.4.2.2.1 General rules

Flash-hook events detected by the line side of the ~~AGCF/~~VGW are reported to the feature manager using a **Feature Request** internal primitive.

For the processing of flash-hook event notifications two different methods exist:

- Loose Coupling Method.
- ~~*Tight Coupling Method.*~~

Note: Loose Coupling Method shall be used.

Among others, the methods determine the degree of autonomous processing in the ~~AGCF/~~VGW after the notification of the flash-hook event to the PES Application Server.

The execution of the service independent feature logic is common for both methods of flash-hook management up to the point in time flash-hook is detected.

Both methods are reflected in separate clauses of clause B.4.2.2 for flash-hook management principles and in annex C for individual supplementary services. The different methods can be characterized as follows:

- Loose Coupling:

  The ~~AGCF/~~VGW implements service-independent feature logic for dealing with register recall events. By evaluating call context and UA profile information, it can make certain decisions such as whether or not to apply *special* dial tone on register recall, collect digits (SOC or SCC), put a party on hold etc. The AS perceives the ~~AGCF/~~VGW almost as if it was a regular IMS client accessing PSTN/~~ISDN~~ simulation services.

- ~~*Tight Coupling:*~~

  ~~*The AS is assumed to have full control over the service. The AGCF/VGW does not implement any service-independent feature logic for dealing with register recall events and does not require UA profile information. The PES AS takes the decisions on the service behaviour (e.g. whether or not to apply dial tone and collect digits), manipulates call legs as needed and instructs the AGCF/VGW on how to proceed based on appropriate SIP messages.*~~

The decision to implement tight coupling or loose coupling is left to the network operator. However, it is important that both the ~~AGCF/~~VGW and the PES application server are configured the same way.

B.4.2.2.2 Loose coupling procedures

Processing of a Feature-Request depends on the call configuration.

~~***Call Configuration: 1-party***~~

~~*In this configuration, an unconfirmed INVITE dialog exists between the SIP UA and another endpoint.*~~

~~*On receipt of a **Feature Request** from the MGC component, unless an explicit indication that the HOLD service is not provisioned to the user has been received as part of the profile delivery procedure, the Feature Manager shall request the MGC component to interact with the media gateway in order to play a dial tone and collect digits.*~~

~~*If the digit collection succeeds, the Feature Manager shall request the SIP UA to send an INVITE request to the PES application server, with the collected digits as Request-URI. Otherwise, the events shall be discarded.*~~

Call Configuration: Stable 2-party

In this configuration, one confirmed INVITE dialog exists between the SIP UA and another endpoint.

On receipt of a flash-hook notification from the MGC component, unless an explicit indication that the HOLD service is not provisioned to the user has been received as part of the profile delivery procedure, the Feature Manager shall:

- Request the SIP UA to send a re-INVITE request towards the connected party (B party). The re-INVITE request is built as follows:
  - The Request URI is set to the B party's identity.
  - The SDP description for the active media stream is set to a=sendonly.

NOTE 1: This information may be used by an AS to request the sending of an announcement to the B Party.

- Request the MGC component to initiate an outgoing call process as if an off-hook event had been received.

Call Configuration: Stable 2-party call with additional held/waiting party

In this configuration, a confirmed INVITE dialog exists between the SIP UA and some other endpoint (the "active party", and either an unconfirmed dialog exists with a third endpoint (the "waiting party"), or another confirmed dialog exists with a third party that is on hold ("held party").

On receipt of a flash-hook notification from the MGC component, the Feature Manager shall request the MGC component to interact with the media gateway in order to:

- Set the stream mode of the IP media termination to "inactive".
- Play a dial tone *(special dial tone).*
- Collect a feature code.

Note: The feature codes “0”, “1”, “2” and “3” are valid depending on the respective service; all others are treaded as wrong codes. (Feature code is the equivalent to Switching Order Command).

The number of digits is provisioned in the ~~*AGCF/*~~VGW. The ~~*AGCF/*~~VGW shall also send a re-INVITE request on the initial dialogue to hold the associated media stream, as described in TS 183 010 [8].

~~*As a Network operator option the*~~*The* sending of a re-INVITE to set the active party to inactive is delayed until the feature logic verifies that the collected feature code is valid.

NOTE: This is also valid when a wrong feature code is received.

NOTE 2: This information may be used by an AS to request the sending of an announcement to the B Party.

Processing of the feature code depends on whether loose or tight coupling procedures are applied between the ~~*AGCF/*~~VGW and the AS. The decision to use tight coupling or loose coupling is left to the network operator. However, it is important that both the ~~*AGCF/*~~VGW and the PES application server are configured the same way.

With loose coupling, the ~~*AGCF/*~~VGW provides call processing logic to manipulate call legs, much like a simulation endpoint would operate.

The Feature Manager analyses the feature code according to a local mapping table. For each feature code, this table indicates the user expected feature:

- The served user wishes to be connected to a particular held/waiting party and keep the other party held/waiting.
- The served user wishes to be connected to a particular held/waiting party and release the other party.
- The served user wishes to establish a 3-party conference with both of the other parties.

- ~~The served user wishes invoke the MCID service.~~

If the feature code received does not match any known feature, ~~the FM ignores the feature code and optionally requests the MGC component to interact with the media gateway in order to play an error tone or an announcement to the served user~~an INVITE containing an indication to play an announcement is sent towards the AS/MRFC/MRF.

When the user pushes Flash-Hook again and no response to the INVITE has been received yet, then a CANCEL shall be sent to withdraw the announcement request.

~~As a network option when~~When an additional Flash-Hook command has been received a new dial tone (special dial tone) is played and feature code can be received by the MGC component and processed by the ~~AGCF/~~VGW. After receiving Hook-Flash at the VGW a "CANCEL" has to be sent towards the AS/MRFC/MRF.

If the feature code received indicates that the served user wishes to be connected to a particular held/waiting party, ~~unless an explicit indication that the Call Toggle service is not provisioned to the user has been received as part of the profile delivery procedure,~~ the Feature Manager shall:

- Request the SIP UA to send either:
  - a 200 OK response to the INVITE request received from the waiting party if the dialogue is not confirmed yet; or
  - a re-INVITE request if the dialogue with the held party is already confirmed. The Re-INVITE request is built as follows:

    The Request URI is set to the held party's identity.

    The SDP description for the active media stream is set to a=sendrecv.

- ~~If a AGCF is in use then request the MGC component to interact with the media gateway in order to:~~
  - ~~modify the Remote Descriptor of the ephemeral termination according to the SDP information received from the held/waiting party;~~
  - ~~set the Stream Mode of the IP media termination to sendrecv;~~
  - ~~monitor the flash-hook event on the physical termination.~~
- If a VGW is used then:
  - modify the IP media termination according to the SDP information received from the held/waiting party;
  - set the Stream Mode of the IP media termination to sendrecv;
  - monitor the flash-hook event on the physical termination.

If the feature code received indicates that the served user wishes to release the call with the other held/waiting party, the Feature Manager shall:

- Request the SIP UA to send a 603 response or a BYE request to the waiting/held party depending on the dialogue state.
- Request the MGC component to interact with the media gateway in order to monitor the flash-hook event on the physical termination.

If the feature code received indicates that the served user wishes to establish a 3-party conference, ~~unless an explicit indication that the 3PTY service is not provisioned to the user has been received as part of the profile delivery procedure,~~ the Feature Manager shall:

- apply the procedures described in Annex C.14.2B.
- ~~Request the MGC component to interact with the media gateway in order to add a termination to the current context based on SDP information associated with the initial held party (i.e. the party that was already held before the flash-hook has been detected).~~

- ~~*Request the SIP UA to send a re-INVITE request towards each of the held/waiting parties. The re-INVITE request is built as follows:*~~
  - ~~*the Request URI is set to the held/waiting party's identity;*~~
  - ~~*The SDP description for the active media stream is set to a=sendrecv. If an AGCF is in use the address and port are set according to the contents of the local descriptor of the termination representing this party on the media gateway. Or if a VGW is in use then the address and port are set according to the SDP description of the IP media termination representing this party.*~~

~~*If the feature code received from the internal MGC component indicates that the served user wishes to invoke MCID, the Feature Manager shall request the SIP UA to send a re-INVITE request towards the AS. The re-INVITE request is built as follows:*~~

- ~~*The Request URI is set to the calling party's identity; and:*~~
  - ~~*include no Body in the re-INVITE; or*~~
  - ~~*as a network operator option a re-INVITE including a XML-MIME with XML mcid body with MCID XML Request schema containing a McidRequestIndicator set to 1.*~~

Delete Section B.4.2.2.3 Tight coupling procedures

B.4.3 Behaviour of Re-ringing

In the following example re-ringing is described:

- Re-ringing after Hold/Waiting:

  Call Configuration: Stable 2 party call with additional held/waiting party.

  Re-ring scenarios require that there is still an existing dialog when the (A-) subscriber has gone on-hook, the active dialog is terminated using BYE and there is still a party on-hold or waiting. The held party will be activated and the activated or the waiting party will be presented to the UE by the re-ringing behaviour. The mechanism for accomplishing this is network specific.

  - *User A goes on-hook;*
  - *VGW sends a BYE to the active call;*
  - *VGW sends ringing towards the user A;*
  - *In case of accepting a waiting call by re-ringing FSK is sent towards user A; in all other cases no FSK is sent towards user A;*
  - *User A goes off hook;*
  - *In case of a waiting call the VGW sends a 200 OK towards the waiting user; or*
  - *In case of a held call the VGW sends a re-INVITE with sendrecv towards the held user.*

Annex C (informative): Implementation of Supplementary Services

C.1 General principles

C.1.1 Introduction

This annex describes guiding principles for implementing commonly deployed PSTN supplementary services using the IMS-based PES architecture. The list of services is taken from EG 201 973-2 [11].

The actual service logic resides in the Application Server and is outside the scope of standardization. This annex focuses on the interactions between the ~~AGCF~~*VGW* and PES application servers. Only the part of the service logic which has an impact on signalling to/from the ~~AGCF~~*VGW* is described in this annex.

Similar procedures may also be used in case of analogue lines connected to a VGW acting as a UE with regard to a P-CSCF in the PES. The present document will describe these procedures of the VGW in the appropriate clause.

~~AGCF involvement in the execution of these services is limited to the generic capabilities described in the main body of the present document for supporting interworking between SIP and H.248 protocols and in annex B for processing flash-hook events.~~

NOTE: The relevant call flows concerning Annex C are provided in Annex B of the main document 1 TR 126.

C.1.2 Supplementary Service control

This annex assumes that subscribers can control their supplementary services using service code commands and switching order commands as defined in ETS 300 738 [12] *and specified in Annex D of 1 TR 114 [Ref_dt1] for Deutsche Telekom*.

~~More than one command can be dialled on a single call. For instance, a caller may dial the "inhibit call waiting" command, followed by the "calling line identity restriction" command, followed by a the called party number.~~

C.1.2.1 Service code commands

Service codes commands are user requests to perform an action that does not result in a call to another party. Actions such as feature activation, feature deactivation, and feature status inquiries are triggered by service code commands.

*Within theNGN platform of Deutsche Telekom the Service Code Commands (SCC) are the *xx#, #xx# sequences to be applied in compliance with Annex D of 1 TR 114 [Ref_dt1].*

C.1.2.1.1 Command syntax

The format of service code commands as defined in ETS 300 738 [12] is reproduced below:

"START PX SC (SR SI) SX" or **"PX SC (SR SI) SX FINISH"**

Where:

- **START** is the start command, e.g. "Off-hook", an alternative to the finish command;
- **PX** is a mandatory service prefix;
- **SC** is a mandatory service code;
- **SR** is one or more separator/s, as required;
- **SI** is one or more units of supplementary information, as required;

- **SX** is a service suffix as required;
- **FINISH** is a finish command, e.g. SEND, an alternative to the start command.

NOTE: Digit maps used by the media gateways or VGW to collect digits, include appropriate alternatives to cope with the syntax of service code commands.

C.1.2.1.2 Generic procedure at the ~~AGCF/~~VGW side

On receipt of a service code command from a media gateway, the ~~AGCF/~~VGW sends an INVITE request to the S-CSCF with the following information:

- A Request-URI structured as follows:
  - A user part containing the service code command, excluding the START and FINISH fields.
  - A domain name which together with the user part provides sufficient information to the S-CSCF to forward the INVITE request to the appropriate AS, based on Initial Filter Criteria stored in the user profile, e.g.:

"PX SC (SR SI) SX"@pes-scc.operator.com

NOTE 1: If the service code command includes a square "#" symbol, the userinfo portion of the Request-URI is in the form of a telephone-subscriber. The series of digits that form the service code command are encoded as a local-number. The phone-context attribute is set to a domain name of the PES operator, e.g. phonecontext=pes-scc.homedomain.com that is specific enough to enable the application server to interpret the commandecode. Setting the phone-context attribute is required for conformance purposes with RFC 3966 [19]. PES network entities (e.g. CSCF) ignore this attribute.

- ~~*An AGCF sends a P-Asserted-Identity header containing the public user identity of the subscriber issuing the service code command or in the case of t*~~*T*he VGW *sends* a P-Preferred-Identity header ~~*may be sent*~~. If the P-Preferred-Identity is not sent then the PES Endpoint ensures that the From header contains the equivalent identity.
- An SDP offer for a voice call.

NOTE 2: The SDP offer may be used by the Application Server in case an announcement has to be delivered.

Delete Section C.1.2.1.3 Generic procedure at the AS side

C.1.2.2 Switching order commands

Switching order commands are typically used to invoke a service or modify the characteristics of a call, such as establish a three-party call.

Within the NGN platform of Deutsche Telekom the Switching Order Commands (SOC) are one digit commands ("0", "1", "2" or "3" depending on the service) to invoke services like HOLD, 3PTY etc.

The format of switching order commands (SOC) is reproduced below:

"START SO (SR SI)" or **"SO (SR SI) FINISH"**

Where:

- **START** is a start command, e.g. "Register Recall (R)", an alternative to the finish command, as required;
- **SO** is a mandatory switching order;
- **SR** is one or more separators, as required;
- **SI** is one or more units of supplementary information, as required;
- **FINISH** is a finish command, e.g. "send", an alternative to the start command, as required.

Processing of switching order commands where the start command is a Register Recall (also known as Flash-Hook event) is described in annex B. ~~Processing of the Register Recall at the AGCF depends on the call configuration and usually involves requesting the media gateway to deliver a dial tone and collect digits.~~

Processing of the Register Recall at the VGW depends on the call configuration and usually the VGW delivers a *special* dial tone and collect*s* digits.

Delete Section C.1.3 Setting of initial filter criteria

C.1.4 Supplementary services using ISUP information

Full support of supplementary services may be realized by exchanging service information between peer SIP signalling entities via SIP signalling and/or encapsulated ISUP information. The ISUP information necessary to support each individual service is specified by the corresponding ETSI or ITU-T supplementary service specification; see table C.1.

Table C.1: Supplementary Service References

Supplementary Service	ETSI/ITU-T Reference
Calling Line Identification Presentation (CLIP)	EN 300 356-3 [34]
Calling Line Identification Restriction (CLIR)	EN 300 356-4 [34]
Connected Line Identification Presentation (COLP)	EN 300 356-5 [34]
Connected Line Identification Restriction (COLR)	EN 300 356-6 [34]
Terminal Portability (TP)	EN 300 356-7 [34]
User-to-User Signalling (UUS)	EN 300 356-8 [34]
Closed User Group (CUG)	EN 300 356-9 [34]
Subaddressing (SUB)	EN 300 356-10 [34]
Malicious Call Identification (MCID)	EN 300 356-11 [34]
Conference Call (CONF)	EN 300 356-12 [34]
Explicit Call Transfer (ECT)	EN 300 356-14 [34]
Call Forwarding Busy (CFB)	EN 300 356-15 [34]
Call Forwarding No Reply (CFNR)	EN 300 356-15 [34]
Call Forwarding Unconditional (CFU)	EN 300 356-15 [34]
Call Deflection (CD)	EN 300 356-15 [34]
Call Hold (HOLD)	EN 300 356-16 [34]
Call Waiting (CW)	EN 300 356-17 [34]
Completion of Calls to Busy Subscriber (CCBS)	EN 300 356-18 [34]
Three-Party (3PTY)	EN 300 356-19 [34]
Completion of Calls on No Reply (CCNR)	EN 300 356-20 [34]
Anonymous Communication Rejection (ACR)	EN 301 798 [39].
Multi-Level Precedence and Pre-emption (MLPP)	ITU-T Recommendation Q.735.3 [35]
Global Virtual Network Service (GVNS)	ITU-T Recommendation Q.735.6 [36]
Reverse charging (REV)	ITU-T Recommendation Q.736.3 [37]

C.2 Advice of Charge

NOTE: The mapping from SIP AOC information to 16kHz metering pulses described below may not be totally applicable since the AOC information to the VGW is sent according to ETSI TS 124 447 [46] and in monetary format.

Delete Section C.2.1 Actions at the Originating AGCF

Delete Section C.2.2 Actions at the Originating AS

Delete Section C.2.3 Actions at the Terminating AGCF

Delete Section C.2.4 Actions at the Terminating AS

C.2.5 Actions at the originating VGW

C.2.5.1 General

When receiving charging pulses from the Application Server (via the S-CSCF) the VGW can either as a network option:

- generate Metering Pulses by one of the methods described in clause C.2.5.2; or
- build an Advice Of Charge message (see EN 200 659-3 [10]) by rules described in annex D and transmit this message.

In addition to the procedures according to ES 283 003 [4], the Originating VGW includes the Accept header field with:

- "application/vnd.etsi.aoc+xml", the MIME type associated with AOC information (see annex E), and indicate the versions of the AOC XML Schema it supports. The versions - of the MIME type associated with AOC information (see annex E) - indicated is corresponding with a value of the version attribute of the <schema> XML element of an AOC XML Schema (see annex E); and
- any other MIME type the served UE is willing and capable to accept.

C.2.5.2 VGW Metering Pulse generation

Delete Section C.2.5.2.1 Usage of an AOC-D for Metering Pulse generation

C.2.5.2.2 Usage of AOC-S for Metering Pulse generation

Annex-E extends the TS 183 047 [7] AOC -XML schema for Pulse Metering usage. This method may be used in order to reduce signalling load between AS and VGW. It does not replace the actual billing information which is stored in CDRs of the responsible Application Servers.

When receiving Pulse Metering Instructions from the PES Application Server (via the S-CSCF), the VGW generates metering pulses.

The type and duration of the pulses to be applied are provisioned in the VGW.

Procedures for Continuous Metering Pulse (Basic Communication continuous charge):

Upon reception of AOC-S for basic communication with continuous type of charging (periodic metering), the VGW generates metering pulses using the Pulse Repetition Interval. The Pulse repetition interval is applied from the XML schema.

Procedures for single Metering Pulse Burst (e.g. for Communication Set-up charge, Service Add-on charge):

Flat rate charging (e.g. at communication set-up or due to add-on charge) requires generation of a metering burst. The Pulse Count is applied from the XML schema and the Pulse repetition interval is provisioned in the VGW.

Procedures for Repetitive Metering Pulse Burst:

The VGW repetitively generates single metering Burst Pulses. The burst repetition interval is applied from the XML schema.

C.2.6 Actions at the terminating VGW

Not applicable.

C.3 Anonymous Call Rejection

Delete Section C.3.1 Actions at the Originating AGCF

Delete Section C.3.2 Actions at the Originating AS

Delete Section C.3.3 Actions at the Terminating AS

Delete Section C.3.4 Actions at the Terminating AGCF

C.3.5 Actions at the Originating VGW

This service does not require the originating VGW to perform any specific action.

C.3.6 Actions at the Terminating VGW

This service does not require the terminating VGW to perform any specific action.

Delete Section C.4 Automatic Call Return

C.5 Calling Line Identity Presentation/Restriction

Delete Section C.5.1 Actions at the Originating AGCF

Delete Section C.5.2 Actions at the Originating AS

Delete Section C.5.3 Actions at the Terminating AS

Delete Section C.5.4 Actions at the Terminating AGCF

C.5.5 Actions at the Originating VGW

A service code command may be received by the VGW in case the calling user wishes to override the default setting for a particular call, in which case the called party number is embedded in the command code as part of the supplementary information. The VGW generates an INVITE request according to the rules described in clause C.1.

Otherwise the originating VGW is not involved in the provision of this service.

C.5.6 Actions at the Terminating VGW

User Identification information received in an INVITE request ("P-Asserted-Id", "Privacy", and "From" headers) is used by the VGW to generate the appropriate Call Setup message (see EN 200 659-3 [10]) to be delivered to the called terminal. Generation of the Call Setup message is further described in annex D.

Delete Section C.6 Calling Name Delivery

C.7 Call Forwarding

Delete Section C.7.1 Activation/Deactivation/Interrogation

C.7.2 Invocation

Delete Section C.7.2.1 Actions at the Originating AGCF

Delete Section C.7.2.2 Actions at the Originating AS

Delete Section C.7.2.3 Actions at the Forwarding AS

Delete Section C.7.2.4 Actions at the Forwarding AGCF

Delete Section C.7.2.5 Actions at the Terminating AS

Delete Section C.7.2.6 Actions at the Terminating AGCF

C.7.2.7 Actions at the Originating VGW

No specific action is performed by the VGW.

C.7.2.8 Actions at the Forwarding VGW

The VGW at the forwarding side is not involved in the processing of forwarded calls.

C.7.2.9 Actions at the Terminating VGW

Call forwarding information received in an INVITE request ("history-info" header) is used by the VGW to generate the appropriate Call Setup message (see EN 200 659-3 [10]) Generation of the Call Setup message is further described in annex D.

Delete Section C.8 Distinctive Ringing

C.9 Call Waiting

C.9.1 General for Loose Coupling

Delete Section C.9.1.1 Actions at the AGCF at the terminating side

Delete Section C.9.1.2 Actions at the AS at the terminating side

C.9.1.3 Actions at the VGW at the terminating side

On receipt of an INVITE request for a busy subscriber, the VGW performs the following actions. ~~unless an explicit indication that Call Waiting service is not provisioned to the user as part of the profile delivery procedure, otherwise the VGW sends a 486 (Busy Here) response towards the Application Server and no action on the analogue line is required. As a Network operator option t~~

*T*he INVITE request to the busy subscriber ~~may~~ include*s* a CW MIME body, in compliance with TS 124 615 [44]:

- VGW apply call waiting tone.
- VGW monitor flash-hook events.
- Send a 180 (Alerting) including an Alert-Info header field set to "urn:service:call-waiting" towards the AS.
- Start a no answer timer.

If the no answer timer expires then ~~dependant on network operator policy~~ the VGW sends a 486 (Busy Here) ~~or 480 (Temporarily unavailable) response towards the Application Server.~~

~~As an option, the calling user could be specifically informed that the called user has not answered the communication if a Reason header field set to cause 19 (no answer from user, user alerted) is included in the 480 (Temporarily unavailable) response.~~

If a flash-hook event is received by the VGW it stops the no-answer timer and requests the media gateway to performthe following actions ~~unless an explicit indication that the HOLD service is not provisioned to the user has been previously received as part of the profile delivery procedure~~:

- ~~Set the stream mode of~~ *Mute* the IP media termination ~~to inactive~~.
- Apply a dial tone *(special dial tone according the main specification 1 TR 126 and Annex B of 1 TR 126)*.
- Collect a switching order command.

~~NOTE 1: In some networks, applying a dial tone and collecting an explicit switching order command is not required as the register recall is interpreted as a request to accept the waiting call. Subsequent VGW procedures are executed as if an equivalent switching order command had been received from the user.~~

The number of digits is provisioned in the VGW. The VGW also sends a re-INVITE request on the initial dialogue to hold the associated media stream, as described in TS 183 010 [8].

Processing of the switching order command depends on whether loose or tight coupling procedures are applied between the VGW and the AS. ~~Figure C.1 illustrates the loose coupling case while figure C.2 illustrates the tight-coupling case.~~

NOTE 2: These figures do not take into account all possible ~~service code~~*switching order* commands that may be received from the user (e.g. ~~service code~~*switching order* command requesting that a waiting call be accepted and the active call be released).

The VGW evaluates the Switch Order Command based on a provisioned mapping table.

Note: The SOC in the network of Deutsche Telekom is one digit long; the possible digits are "0", "1" or "2".

If the SOC indicates that the served user wishes to be connected to the waiting party, the VGW performs the following actions:

- *The VGW also sends a re-INVITE request on the initial dialogue to hold the associated media stream, as described in TS 183 010 [8].*
- Send a 200 OK response to the INVITE request received from the waiting party.
- The VGW procedures are:
  - Modify the IP media termination according to the SDP information received from the waiting party.
  - Monitor the flash-hook event.

If SOC indicates that the served user wishes to reject the waiting call, the VGW performs the following actions:

- Send a provisioned error response code (e.g. 603) to the INVITE request received from the waiting party.
- The VGW.
  - Set the stream mode of the IP media termination to send-receive.
  - Monitor the flash-hook event.
- ~~*Send a re-INVITE request towards the held party (i.e. the party that has been held for the purpose of collecting the switching order command). The re-INVITE request is built as follows:*~~
  - ~~*The Request-URI is set to the held party's identity.*~~
  - ~~*The SDP description for the active media stream is set to a=sendrecv.*~~

Once the communication is established with the waiting party, the user may decide to switch back to the initial party, using a Register Recall followed by new switching order command. If the value of the switching order command indicates that the initial party is switched back, the VGW performs the following actions:

- *Send a re-INVITE request towards the active party. The re-INVITE request is built as follows:*
  - *The Request URI is set to the active party's identity.*
  - *The SDP description for the active media stream is set to a=sendonly.*
- Send a re-INVITE request towards the held party. The re-INVITE request is built as follows:
  - The Request-URI is set to the held party's identity.
  - The SDP description for the active media stream is set to a=sendrecv.
- ~~*Send a re-INVITE request towards the active party. The re-INVITE request is built as follows:*~~
  - ~~*The Request URI is set to the active party's identity.*~~
  - ~~*The SDP description for the active media stream is set to a=sendonly.*~~
- The VGW:
  - Modify the IP media termination according to the SDP information associated with the held party.
  - Monitor the flash-hook event.

If the feature code received does not match any known feature, the VGW ignores the feature code and optionally play an error tone or an announcement to the served user..

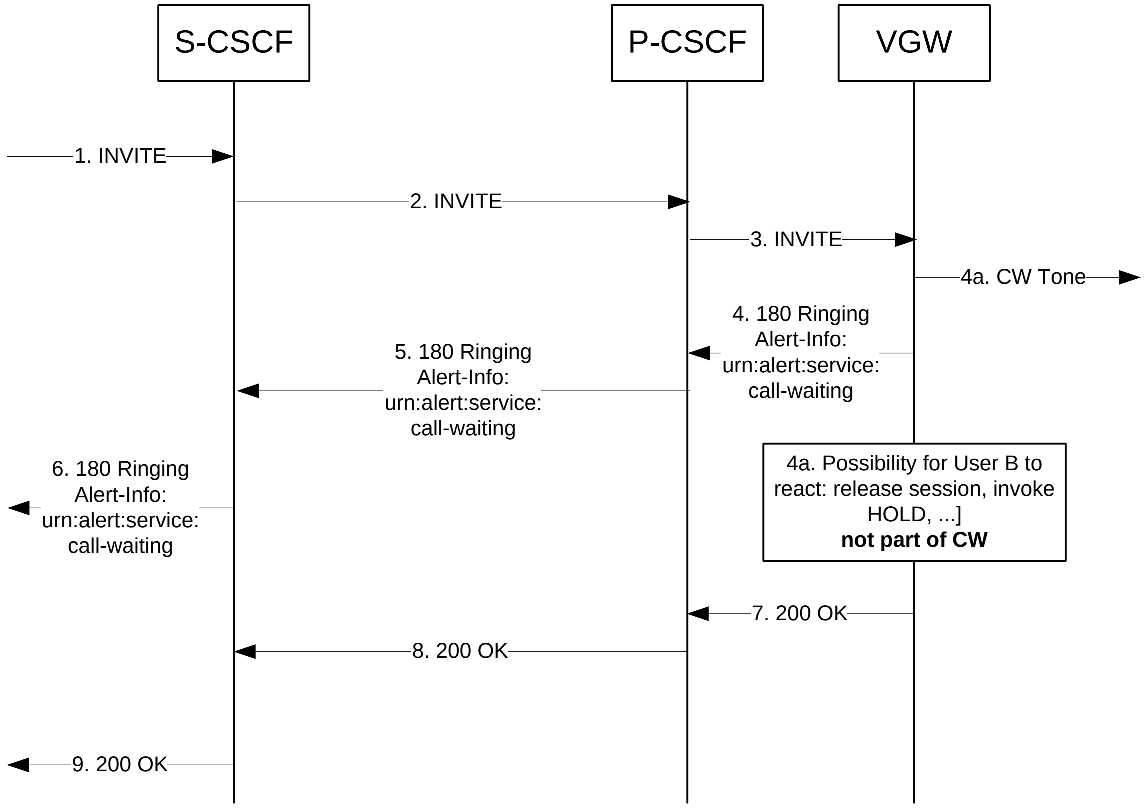

Figure C.1A: Call Waiting with loose VGW coupling ***(modified for DT use case)***

C.9.2 Void

C.9.2.1 Void

C.9.2.2 Void

Delete C.9.3 General for Tight coupling

C.10 Incoming Call Barring

C.10.1 Activation/Deactivation/Interrogation

Delete Section C.10.1.1 Actions at the AGCF

Delete Section C.10.1.2 Actions at the AS

C.10.1.3 Actions at the VGW

Activation, deactivation and interrogation of incoming call barring is performed using service code commands as described in ETS 300 738 [12] *and specified in Annex D of 1 TR 114 [Ref_dt1] for Deutsche Telekom*. On receipt of a service code command, the ~~*AGCF*~~*VGW* sends an INVITE request as described in clause C.1.

C.10.2 Invocation

Delete Section C.10.2.1 Actions at the Originating AGCF

Delete Section C.10.2.2 Actions at the Originating AS

Delete Section C.10.2.3 Actions at the Terminating AS

Delete Section C.10.2.4 Actions at the Terminating AGCF

C.10.2.5 Actions at the Originating VGW

The originating VGW is not involved in the invocation of the service.

C.10.2.6 Actions at the Terminating VGW

The terminating VGW is not involved in the provision of the service.

C.11 Malicious Call Identification (Loose coupling)

Delete section C.11.1 Actions at the Originating AGCF

Delete section C.11.2 Actions at the Originating AS

Delete section C.11.3 Actions at the Terminating AS

Delete section C.11.4 Actions at the Terminating AGCF

C.11.5 Actions at the Originating VGW

The VGW at the originating side is not involved in the provision of this service.

C.11.6 Actions at the Terminating VGW

~~*As a network option, the VGW may use the annex A Profile Delivery mechanism for obtaining the subscriber MCID subscription status (provisioned/withdrawn).*~~

~~*In case of invoking the MCID via a special service code command sent from the user the VGW sends a re-INVITE without any body.*~~

~~*As a network operator option a re-INVITE including a XML-MIME with XML mcid body with MCID XML Request schema containing a McidRequestIndicator set to 1 may be sent.*~~

Note: MCID is invoked by Service Code Commands only or is provided permanently.

Delete Section C.11A Malicious Call Identification (Tight coupling)

C.12 Message Waiting Indicator

Delete section C.12.1 Actions at the AGCF

Delete section C.12.2 Actions at the AS

C.12.3 Actions at the VGW

On receipt of a NOTIFY request reporting the "message-summary" event, the VGW:

- Modify the default dial tone.

Send a Message Waiting Indicator message (see EN 200 659-3 [10]).

C.13 Outgoing Call Barring

C.13.1 Activation/Deactivation/Interrogation

Delete section C.13.1.1 Actions at the AGCF

Delete section C.13.1.2 Actions at the AS

C.13.1.3 Actions at the VGW

Activation and deactivation of outgoing call barring is performed using service code commands as described in ETS 300 738 [12] *and specified in Annex D of 1 TR 114 [Ref_dt1] for Deutsche Telekom*. On receipt of a service code command, the VGW sends an INVITE request as described in clause C.1.

C.13.2 Invocation

Delete section C.13.2.1 Actions at the Originating AGCF

Delete section C.13.2.2 Actions at the Originating AS

Delete section C.13.2.3 Actions at the Terminating AS

Delete section C.13.2.4 Actions at the Terminating AGCF

C.13.2.5 Actions at the Originating VGW

The originating VGW is not involved in the invocation of the service.

C.13.2.6 Actions at the Terminating VGW

The terminating VGW is not involved in the provision of the service.

C.14 Three Party Service

C.14.0 General

~~*Processing of the switching order command depends on whether loose or tight coupling procedures are applied between the AGCF and the AS. Moreover, in loose coupling case, two methods can be used: the INVITE method, described in clause C.14.2, and the REFER method, described in clause C.14.2A. Figure C.4 illustrates the loose coupling case with the INVITE method while figure C.5 illustrates the loose coupling case with the REFER method. The tight coupling case is described in clause C.14.3.*~~

A networkbased solution for Three Party Service is currently not supported.

AS functionality includingannouncements shapp be provided by the VGW

C.14.1 General for Loosely Coupled Options

Delete Section C.14.1.1 Actions at the AGCF at the service invocation side

Delete Section C.14.1.2 Actions at the AS at the service invocation side

C.14.1.3 Actions at the VGW at the service invocation side

The service is invoked during a stable 2 party call (without any waiting/held call) using the Register Recall. ~~*Figure C.3 illustrates the message sequence between the VGW and the AS.*~~

On receipt of a flash-hook event, the VGW performs the following actions unless an explicit indication that the HOLD service is not provisioned to the user has been previously received as part of the profile delivery procedure:

- Play a dial tone on the physical and collect digits.
- Send a re-INVITE request to place the current call on hold.

On receipt of the dialled digits, the VGW opens a new dialogue by sending an INVITE request with the following elements:

- The dialled digits used as a Request-URI.
- An SDP Offer for a voice call.

The VGW then perform the following actions:

- monitor the flash-hook event;
- set the stream-mode of the IP media termination to "inactive"; and
- waits for an incoming SIP message.

On receipt of 180 (Ringing) without P-Early-Media header or with a P-Early-Media header set to a value different from "sendonly" or from "sendreceive", the VGW performs the following actions:

- Play a ringback tone.

On receipt of 180 (Ringing) or 183 (Session Progress) with a P-Early-Media header set to "sendonly" or "sendreceive", the VGW performs the following actions:

- Modify the configuration of the IP media termination so as to ensure that the end user will perceive early media.

On receipt of an SDP Answer in a 200 (OK) or in one of the above provisional responses, the VGW performs the following actions:

- Modify the IP media termination associated with the physical termination representing the analogue line.

On receipt of a re-INVITE request requesting a call to be placed on hold, the AS applies the procedures described in TS 183 010 [8].

~~*Processing of the switching order command depends on whether loose or tight coupling procedures are applied between the VGW and the AS. Moreover, in loose coupling case, two methods can be used: with the INVITE method, described in clause C.14.2, and with the REFER method, described in clause C.14.2A. Figure C.4 illustrates the loose coupling case with the INVITE method while figure C.5 illustrates the loose coupling case with the REFER method. The tight coupling case is described in clause C.14.3.*~~

Delete section C.14.2 Loose coupling Option1 with INVITE method

C.14.2B Loose coupling Option 3 by sending INVITE request with URI list

Delete section C.14.2B.1 Actions at the AGCF at the invoking side

Delete section C.14.2B.2 Actions at the Originating AS at the invoking side

C.14.2B.3 Actions at the VGW at the invoking side

The ~~AGCF~~VGW evaluates the Switch Order Command based on a provisioned mapping table.

If the feature code received does not match any known feature, the ~~AGCF~~VGW ignores the feature code and ~~optionally to play an error tone or an announcement to the served user. Or~~ initiate an announcement due to the procedures described within TS 183 028 [6].

As a network option the ~~A~~VGW can evaluate a Switch Order Command based on a provisioned mapping table to correct a preceding Switch Order Command that was wrong.

If the SOC indicates that the user wishes to establish a 3-party conference with the held parties the ~~AGCF~~VGW performs the following actions ~~unless an explicit indication that the 3PTY service is not provisioned to the user has been previously received as part of the profile delivery procedure:~~

VGW creates a conference in building the 3PTY context withinthe MG functionality within the VGW. ~~and invites user B and user C to the conference by sending an INVITE to the Conference Factory URI and including URI list in the INVITE request, VGW indicates the certain dialogs which be re-used for this conference in the uri list by ? mechanism.~~

```
INVITE CONF AS
To: CONF AS
From: A
Require: recipient-list-invite

Content-Type: application/resource-lists+xml
Content-Disposition: recipient-list

<?xml version="1.0" encoding="UTF-8"?>
 <resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"
     xmlns:cp="urn:ietf:params:xml:ns:copyControl">
  <list>
   <entry uri="B?Call-ID=1a&From=A%3Btag%3Da&To=B%3Btag%3Db" cp:copyControl="to"/>
   <entry uri="C?Call-ID=2a&From=A%3Btag%3Da&To=C%3Btag%3Dc" cp:copyControl="to"/>
  </list>
 </resource-lists>
```

Once the 3-party call is established with the waiting party, processing of the flash-hook event is similar to the call waiting service. If SOC indicates that the served user wishes to reject one of the parties, the ~~AGCF~~VGW performs the following actions:

- ~~Send a BYE request towards the Conference Server.~~
- Send a BYE request towards on the held dialog of the party that should be disconnected.
- Send a Re-INVITE on the other held party to reactivate that dialog.
- Monitor the flash-hook event.

Delete section C.14.3 General for Tight coupling

Delete section C.15 Repeat Last Call

C.16 Call Hold

C.16.1 Option 1 (Loose Coupling)

Delete section C.16.1.1 Actions at the AGCF at the service invocation side

Delete section C.16.1.2 Actions at the AS at the service invocation side

C.16.1.3 Actions at the VGW at the service invocation side

The service is invoked during a stable 2 party call using the Register Recall.

The VGW monitors the flash-hook event ~~*unless an explicit indication that the Hold service is not provisioned to the user has been previously received as part of the profile delivery procedure*~~.

On receipt of ~~*a NOTIFY request reporting*~~ a flash hook event, the VGW performs the following actions:

- Monitor the flash-hook event.
- Send a re-INVITE request to place the current call on hold.

On receipt of ~~*a NOTIFY request reporting a*~~ flash hook event, the VGW performs the following actions:

- *Play special dial tone and collect a switching order command (SOC); or*
- *Go on-hook and wait for Re-Ringing*
- Send a re-INVITE request to resume the call on hold.
- Monitor the flash-hook event.

On receipt of a re-INVITE request requesting a call to be placed on hold, the AS applies the procedures described in TS 183 010 [8].

~~*As a network option t*~~*T*he AS of the invoking UE initiates the procedures for the provision of an announcement to the held user in accordance with TS 183 028 [6].

Delete section C.16.2 Option 2 (Tight Coupling)

C.17 Call Toggle/Broker Call Service

C.17.1 General

Delete section C.17.1.1 Actions at the AGCF at the service invocation side

C.17.1.2 Actions at the VGW at the service invocation side

The service is invoked having an active call and a held call using the Register Recall.

The VGW monitors the flash-hook event.

C.17.2 Option 1 (Loose coupling)

Delete section C.17.2.1 Actions at the AGCF

C.17.2.3 Actions at the VGW

On receipt of a ~~*NOTIFY request reporting a*~~ flash hook event, the VGW performs the following actions:

- ~~*Set the stream mode of the IP media termination to inactive.*~~
- Apply a dial tone.
- Collect a switching order command.

The VGW evaluates the Switch Order Command based on a provisioned mapping table.

If the feature code received does not match any known feature, the VGW ignores the feature code and ~~*optionally play an error tone or an announcement to the served*~~.

If the SOC indicates that the user wishes to toggle the active and the held parties the VGW performs the following actions ~~*unless an explicit indication that the Call Toggle service is not provisioned for this user has been previously received as part of the profile delivery procedure*~~:

- Send a re-INVITE request towards the active party. The re-INVITE request is built as follows:
    - The Request URI is set to the active party's identity.
    - The SDP description for the active media stream is set to a=sendonly.
- Send a re-INVITE request towards the former held party. The re-INVITE request is built as follows:
    - The Request-URI is set to the held party's identity.
    - The SDP description for the active media stream is set to a=sendrecv.
- The VGW perform the following procedures:
    - Modify the IP media termination according to the SDP information associated with the held party.
    - Monitor the flash-hook event.

Delete section C.17.3 Option 2 (Tight coupling)

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

D.1 Introduction

This annex describes the mapping between SIP messages received by ~~an AGCF or~~ a VGW acting as a UE and the messages of the subscriber line protocol defined in ES 200 659-3 [10]. *For detailed information see also 1 TR 110-1 [Ref_dt3], Anhang B.*

D.2 Call Setup message

This message is used to send information related to an incoming call. e.g. Calling Line Identification Presentation (CLIP) and related services.

This message is built by ~~the AGCF or~~ the VGW on receipt of an initial INVITE request.

~~*The AGCF requests the media gateways to send this message over the analogue line using the Display Data Block parameter of the Display With Alerting signal of the **andisp** package defined in ITU-T Recommendation H.248.23 [13].*~~

The ~~AGCF or~~ VGW populates the message parameters as described in table D.1, based on the contents of the INVITE message and local configuration data.

Table D.1: Call set-up message parameters *(terminating at the VGW)*

Parameter type	O/M	Populating rules
Date and Time	~~O~~ n/a	Set from local clock (see note 1)
Calling Line Identity or Reason for absence of Calling Line Identity	M	Set according to the contents of the "From" header ~~or "P-Asserted-Identity" header dependent on national operator requirements~~. The leading "+" of a received ID shall be replaced by "00". Or ~~Set from "Privacy" header (see note 2).~~
~~Called Line Identity~~	~~O~~	~~"P-Called-Party-Id" header.~~
~~Calling Party Name~~ ~~or~~ ~~Reason for absence of Calling Party Name~~	~~O~~	~~Set according to the "Display-Name" in the "From" header or "P-Asserted-Identity" header dependent on national operator requirements~~ ~~Or~~ ~~"Privacy" header (see note 2).~~
~~Complementary Calling Line Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
Call type	O	Setting of this parameter is based on operator specific rules.
~~First Called Line Identity~~	~~O~~	~~Set according to the "hi-targeted-to-uri " in the first entry in the "History-Info" header~~ ~~Absent if the "Privacy" header set to "history".~~
~~Number of Messages~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
Type of Forwarded call	O	~~Set according to the "Cause" parameter associated with the "hi-targeted-to-uri" in the last entry of the "History-Indo" header.~~ ~~Absent if the "Privacy" header is set to "history".~~
~~Type of Calling User~~	~~O~~	~~Set from the cpc parameter of the P-Asserted-Identity header~~ ~~Absent if the "Privacy" header set to "header".~~
~~Redirecting Number~~	~~O~~	~~Set according to the "hi-targeted-to-uri" in the last entry in the "History-Info"~~ ~~Absent if the "Privacy" header set to "history".~~
~~Network Provider Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Carrier Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Selection of Terminal Function~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Display Information~~	~~O~~	~~May be set from the contents of the message body.~~
~~Service Information~~	~~O~~	~~May be set from the contents of the message body.~~

Parameter type	O/M	Populating rules
~~Extension for network operator use~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
NOTE 1: The ~~AGCF and the~~ VGW shall support an appropriate clock synchronization mechanism. NOTE 2: If the "Privacy" header is included and set to "id" or "user" or "header", the Reason for Absence is set to "Private" (0101 0000). If the "P-Asserted-Identity" header and the "From" header are absent, the Reason for Absence is set to "unavailable" (0100 1111).		

D.3 Message Waiting Indicator message

This message type is used to handle information related to messages in a message system.

This message is built by ~~the AGCF or~~ the VGW on receipt of a NOTIFY request reporting the "message-summary" event.

~~*The AGCF requests the media gateways to send this message over the analogue line using the Data Block parameter of the Generic Data Signalling signal of the **andisp** package defined in ITU-T Recommendation H.248.23 [13]. The value of the TAS parameter is provisioned in the AGCF or MGF, per media gateway or per line.*~~

The ~~AGCF or the~~ VGW populates the message parameters as described in table D.2, using the information contained in NOTIFY requests reporting the "message-summary" event and local configuration data.

Table D.2: Message Waiting Indicator message parameters

Parameter type	O/M	Populating rules
Date and Time	O	Set from local clock (see note).
Calling Line Identity or Reason for absence of Calling Line Identity	O	Set according to the contents of the "P-Asserted-Id" header Or Privacy header.
Calling Party Name or Reason for absence of Calling Party Name	O	Set according to the "P-Asserted-Id" header Or "Privacy" header.
Visual Indicator	M	Set to "FF"H if the "Messages-Waiting" header is set to "yes".
Message Identification	O	Set according to the "Message-ID" header.
Last Message CLI	O	"From" header associated with the last message.
Complementary Date and Time	O	"Date" header associated with the last message.
Complementary Calling Line Identity	O	Setting of this parameter is based on operator specific rules.
Number of Messages	O	Set according to the "Voice-Message" header.
Type of Calling User	O	Setting of this parameter is based on operator specific rules.
Network Provider Identity	O	Setting of this parameter is based on operator specific rules.
Selection of Terminal Function	O	Setting of this parameter is based on operator specific rules.
Display Information	O	May be set from the contents of the message body.
Extension for network operator use	O	Setting of this parameter is based on operator specific rules.
NOTE: The ~~AGCF and the~~ UE shall support an appropriate clock synchronization mechanism.		

D.4 Advice of Charge message

Only pulse metering with a frequency of 16-kHz is used.

***Special case for tariff changes**: If the "transitioning behaviour" is set to "continues" and a tariff sequence without pauses between individual pulses or bursts occurs, the VGW shall apply the "immediate" mode.*

NOTE: The mapping from SIP AOC information to 16kHz metering pulses may not be totally applicable since the AOC information to the VGW is sent according to ETSI TS 124 447 [46] and in monetary format.

~~*This message is used to send information related to the charge of a call.*~~

~~This message is built by the AGCF or the VGW on receipt of a SIP message that contain information defined in TS 183 047 [7].~~

~~The AGCF or the VGW populates the message parameters as described in table D.3, based on information defined in TS 183 047 [7] and local configuration data.~~

~~Table D.3: Advice Of Charge message parameters~~

~~Parameter type~~	~~O/M~~	~~Populated From~~
~~Date and Time~~	~~O~~	~~Set from local clock (see note).~~
~~Calling Line Identity~~ ~~or~~ ~~Reason for absence of Calling Line Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Called line identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Complementary Calling Line Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Charge~~	~~M~~	~~Set from the "recorded-charges" element defined in TS 183 047 [7].~~
~~Additional Charge~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Duration of the call~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Network Provider Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Carrier Identity~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Selection of Terminal Function~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~Display information~~	~~O~~	~~Set from the value of the "billing-id" element defined in TS 183 047 [7].~~
~~Extension for network operator use~~	~~O~~	~~Setting of this parameter is based on operator specific rules.~~
~~NOTE: The AGCF and the VGW shall support an appropriate clock synchronization mechanism.~~		

Delete section Annex E (normative):

Annex F (normative): Overlap Sending

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

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~~

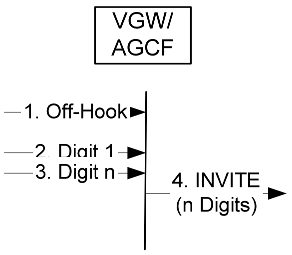

Figure F.1-*1*: Receipt of Digit information at the originating VGW~~/AGCF~~

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 case 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 mandatory.

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.

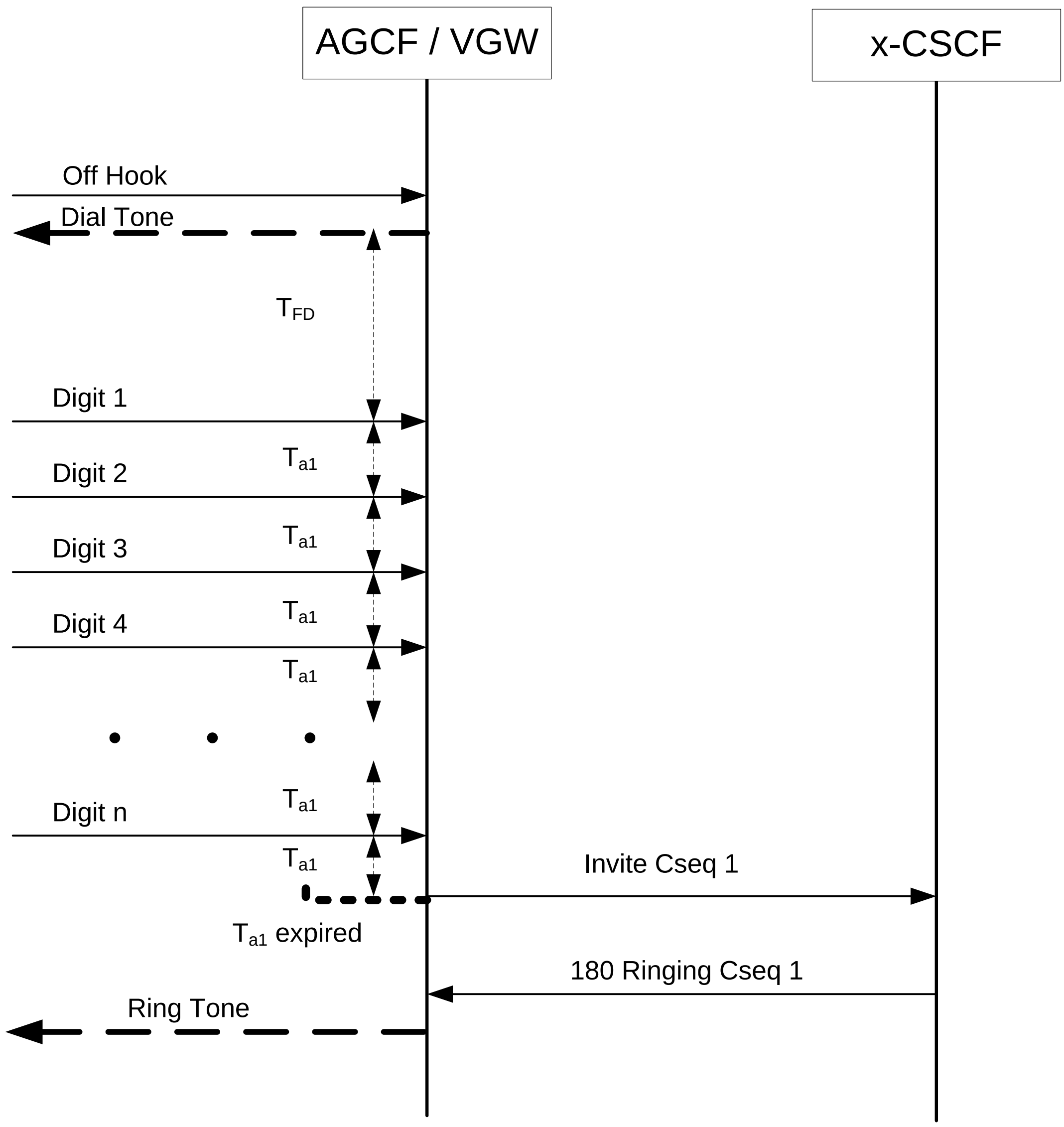

Figure F.1-2: Overlap with Ta1

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.~~				

Delets 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

Date	WG Doc.	CR	Rev	CAT	Title / Comment	Current Version	New Version
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
20-01-10	DTAG				1TR126 Annex A: Modifications incorporated for Deutsche Telekom, only !	0.0.0	0.0.1
16-02-10	DTAG				1TR126 Annex A: Clause C.9.1.3: SCO “3” (HOLD) deleted.	0.0.1	0.0.2
24-02-10	DTAG				1TR126 Annex A: All clauses/text referring to AGCF deleted; Content corrected.	0.0.2	0.0.3
25-03-10	DTAG				1TR126 Annex A: Formal corrections; Document properties updated.	0.0.3	0.0.4
01-06-10	DTAG				1TR126 Annex A: ZTE replaced by FME D; Ref.: T-Home replaced by Deutsche Telekom; Document properties updated.	0.0.4	0.0.5
21-06-10	DTAG				1TR126 Annex A: Figure C.6 corrected (Session-ID); Clause C.14.2B.3: Paragraph “INVITE CONF AS” replaced by the correct one; Clause C.1.1: Note added; All references to deleted call flows deleted.	0.0.5	0.0.6
30-06-10	**DTAG**				**First version of 1 TR 126: Annex A; Modified version for VGW (IAD) connected to accesses of Deutsche Telekom only !**	**0.0.6**	**1.0.0**

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 126: Annex A;** 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 1TR126 are taken out and incorporated in the present document; - All clauses/text referring to AGCF deleted; - All references to deleted call flows deleted; - Clause C.1.1: Note added; - Figure C.6 corrected (Session-ID); - Clause C.9.1.3 (Note): SCO "3" (Hold) deleted; - C.14.2B.3: Paragraph "INVITE CONF AS" replaced by the correct one; - Content corrected; - Document properties updated; - Zentrum Technik Einführung replaced by Fixed Mobile Engineering Deutschland; - T-Home replaced by Deutsche Telekom.
V1.1.0	**December 2011**	**Updated edition of 1 TR 126: Annex A** - Editorial, replace striken Sections by indtructzions Delete Section x.y.z - CONF shall apply within the VGW - HOLD and Call Wait only end device solution