Technical Specification for SIP User Equipments (UE) providing IMS simulation services via analogue (POTS) interfaces (POTS/SIP interworking) using the NGN platform of Deutsche Telekom

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Contents

Foreword

This Technical Specification (Technische Richtlinie, TR) has been produced by the department FMED 15of Deutsche Telekom Netzproduktion GmbH, Fixed Mobile Engineering Deutschland (in the following named as Deutsche Telekom) and it defines the simulation services for IP Multimedia Subsystem (IMS) provided by a SIP User Equipment (UE) via an analogue port (AnPort).

Annex A of the present document is a delta specification based on the ETSI Technical Specification TS 183 043 [8] (endorsement).

1 Scope

The present Technical Specification (TR) is applicable to SIP User Equipments (UE) providing an analogue interface (e.g. Integrated Access Device: IAD) to be connected to an IP access of Deutsche Telekom. It describes the interworking requirements between this analogue interface and the Session Initiation Protocol (SIP) / Session Description Protocol (SDP) for services provided by the Next Generation Network (NGN) of Deutsche Telekom according to the AGB [1] of Deutsche Telekom.

The present document does not describe the physical characteristics and transmission requirements neither of the analogue interface nor the IP access.
A possible physical access is e.g. an xDSL interface provided by Deutsche Telekom which is described in the technical specification 1TR112 [3]. The SIP interface is described in the technical specification 1TR114 [4]. The technical requirements of the AnPort shall guarantee full functionality of TEs, according to 1TR110-1 [2].

Figure 1-1 depicts the scope of the relevant technical specifications.

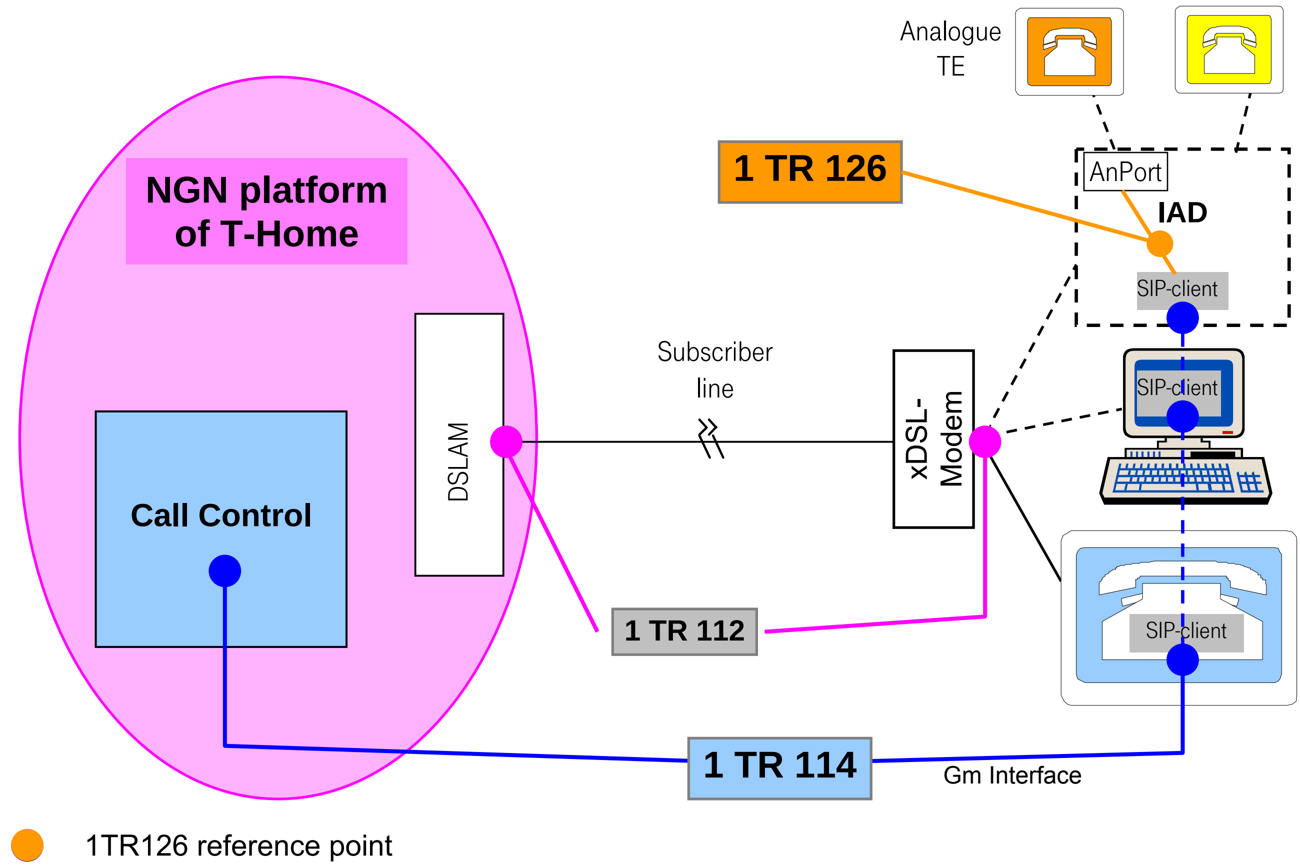


Figure 1-1: Scope of the relevant technical specifications

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- 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.
- For a non-specific reference, the latest version including amendments, errata and corrigenda applies.
- Date of publication in square brackets [ ] refer just to the last known version while this document was in revision.

[1] AGB: Allgemeine Geschäftsbedingungen der Deutsche Telekom (see: www.telekom.de/agb)

[2] T-Com 1TR110-1: Technische Beschreibung der Analogen Wählanschlüsse am T-Net/ISDN der T-Com; Telefonanschlüsse ohne Durchwahl.

[3] T-Home 1TR112: Technical Specification of the U-Interfaces of xDSL Systems in the network of T-Home.

[4] DT 1TR114: Technical Specification of the VoIP service interface between the User Equipment (UE) and the VoIP platform of Deutsche Telekom.

[5] ETSI TS 124 410 V8.0.0 (2008-04): Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; NGN Signalling Control Protocol; Communication HOLD (HOLD) PSTN/ISDN simulation services; Protocol specification (3GPP TS 24.410 version 8.0.0 Release 8)

[6] 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)

[7] ETSI TS 183 036 V2.1.1 (2009-01): Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification.

[8] ETSI TS 183 043 V2.3.1 (2009-03): Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS - based PSTN/ISDN Emulation; Stage 3 specification.

[9] ETSI EN 300 659-1 V1.3.1 (2001-01): 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 1: On-hook data transmission.

[10] ETSI EN 300 659-2 V1.3.1 (2001-01): 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 2: Off-hook data transmission.

[11] ETSI ES 200 659-3 V1.4.1 (2004-08): 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.

[12] IETF RFC 4733: RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals; December 2007.

3 Abbreviations, Definitions and symbols

Abbreviations, definitions and symbols, not listed hereafter, are defined in the reference documents in clause 2.

Abbreviations

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

-1-	
3GPP	Third Generation Partnership Project
3PTY	Three Party Conference
-A-	
AOC	Advice Of Charge
AS	Application Server
AGCF	Access Gateway Control Function
-B-	
-C-	
CC	Call Control
CGP	Charging Determination Point
CgPN	Calling Party Number
CdPN	Called Party Number
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CW	Call Waiting
-D-	
DSS1	Digital Subscriber System No 1
DT	Deutsche Telekom
-E-	
ES	European Standard
ETSI	European Telecommunication Standardisation Institute
ETSI ES	ETSI Standard (normative)
ETSI TR	ETSI Technical Report (informative)
ETSI TS	ETSI Technical Specification (normative)
-F-	
FSK	Frequency Shift Keying
-G-	
-H-	
-I-	
IAD	Integrated Access Device
IP	Internet Protocol
IMS	IP Multimedia Subsystem
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union, Telecommunication Branch
-J-	
-K-	
-L-	
-M-	
MG	Media Gateway
MGC	Media Gateway Controller

MRF	Media Resource Function
-N-	
NSS	Narrowband Signalling Syntax
NGN	Next Generation Networks
-O-	
-P-	
PES	PSTN Emulation Subsystem
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
-Q-	
-R-	
-S-	
SCC	Service Command Code
SOC	Switching Order Commands
SDP	Session Description Protocol
SIP	Session Initiation Protocol
-T-	
TE	Terminal Equipment
TR	Technical Recommendation
TR	Technical Report [ETSI]
TS	Technical Specification
TS	Technical Specification [ETSI]
-U-	
UA	User Agent
UE	User Equipment
ULS	Unnecessary Line Seizure (state)
URI	Universal Resource Identifier
USS	Unnötiger Schleifenschluss
-V-	
VGW	VoIP Gateway (e.g. IAD)
VoIP	Voice over IP
-W-	
-X-	
xDSL	x Digital Subscriber Line (x stands for various kinds of bit rates)
XML	Extensible Markup Language
-Y-	
-Z-	

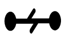
Definitions

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

Term	Definition / Remark
User Equipment	Any device (terminal) at the subscriber premises used by an end user to communicate. It can be e.g. an IAD or telephone set, or any other telecommunication device.
Terminal Equipment	Any device (terminal) at the subscriber premises used by an end user to communicate. It can be e.g. a telephone set, fax machine or any other telecommunication device.
IAD	A user equipment at the subscriber premises which provides different kinds of interfaces (e.g. analogue ports (POTS) and/or ISDN ports (DSS1, So), etc.) for VoIP services; it handles the interworking between those interfaces and the SIP interface. An IAD can be seen as a VGW.
User Agent	A client application used with a particular network protocol, such as Session Initiation Protocol (SIP); it refers to both end points of a phone call, server and client.
Call Control	In telephony, call control refers to the software within a telephone switch that supplies its central function. Call control decodes addressing information and routes telephone calls from one end point to another. It also creates the features that can be used to adapt standard switch operation to the needs of users. Call control software, because of its central place in the operation of the telephone network, is marked by both complexity and reliability.
NGN platform (of Deutsche Telekom)	The entirety of central servers and gateways, as well as software within a telephone network which provides telephony services.
ULS	The Unnecessary Line Seizure state is equivalent to the routine “USS” (Unnötiger Schleifenschluss) as described in [2] used in the network of Deutsche Telekom. Transitions into this ULS state happens in error cases or after certain timers have expired. During this state neither an outgoing nor an incoming communication is possible at the relevant analogue port. To leave this state the TE has to go into the on-hook state for a certain time (e.g. 2s).

Symbols

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

Symbol	Definition
Hz	Hertz
kHz	Kilo hertz
ms	Milliseconds
s	Seconds
t	Time
T_{FD}	Timer "First Digit"
T_{IDP}	Timer "Inter-Digital Pause" (> 500ms): Time interval between two breaks belonging to two consecutive digits where the pulsing loop is closed.
T_{BT}	Timer "Busy Tone"
T_{RT}	Timer "Ringing Tone"
T_{Ring}	Timer "Ringing"
T_{RRing}	Timer "Re-ringing"
T_{a1}	Timer "Timeout 1" (max. pause between dialling digits)
$T_{Retention}$	Timer "Retention" (network timer)
T_{CCxx}	Timer "CCBS", "CCNR" and "CCNL" (network timer)
*	Star sign
#	Hash sign
	Pick-up the receiver (Off-hook)
	Hang-up the receiver (On-hook)
	Hook Flash (flash hook function) (procedure to invoke "register recall"); In case of a SIP UE this procedure shall invoke an equivalent function by sending the relevant SIP message according to ETSI TS 124 410 [5].
< >	Information in pointed brackets is an obliged variable input (signs or figures)

4 Interworking requirements (POTS-SIP)

4.1 Declarative statements

- In the present document the used states P1 to P5.3 refer to the mapping instance associated to the relevant port. The states of the analogue interfaces refer explicitly to the technical specification 1TR110-1 [2].
- The physical interface of an analogue port (DC and AC conditions) shall be defined properly so that terminal equipments which are conform to 1TR110-1 [2] work correctly.
- In the present document the states of an analogue port do consider only one end-to-end communication. The states concerning further connections (e.g. during HOLD) are not taken into account.
- In the following, only those headers are mentioned which are necessary for signalling information. In general there are no ACK-Requests specified, neither in the main body nor in the annexes of the present document.
- A BYE request shall be interpreted as a termination of the previous request, even before establishing a dialogue.
- The potentials for line monitoring and line feeding shall be provided by the VGW as well as audible tones shall be provided by the VGW according to the port state and the associated profile.
- Audible tones shall be aborted after timeout or stopping of the relevant supervisory timer (T_{BT} or T_{RT}).
- The VGW shall support the Dual-Tone-Multifrequency signalling (DTMF).
- The VGW may optionally support pulse dialling; if this option is used all relevant dialling procedures have to be considered in the present document.
  In case of changing the dialling mode (from pulse dialling to multifrequency signalling or vice-versa) during the dialling state by the TE (subscriber) the VGW shall either
  - reject this process by sending an error information (announcement or congestion tone) and skip into state P5.3, or
  - recognize and accept the change of the dialling mode and proceed the communication set-up in a normal way.

  NOTE: Changing the dialling mode from pulse dialling to multifrequency signalling is permitted during the communication state.
- The received INVITE-Request information in the From-Header shall be mapped unvalued to FSK signalling.
- For the reduction of echoes caused by analogue lines/POTs connected to the VGW (IAD) echo cancellation according to Recommendation ITU-T G.168 should be applied. Due to time delay and dispersion time the tail capacity of an echo canceller should be at least 32 ms. In the presence of residual acoustic echo it should be ensured that the performance of an echo canceller with electrical echo is not overly degraded.

- In the present document addresses are described in the general form as follows:

pUserID@domain	Analogue port address in the VGW
CgPN@domain	Originating address of a connection
CdPN@domain	Destination address or dialled number
cc@domain	Call Control address
adrinfo@domain	Digits of the dialled number
illegal_digit@domain	Indication for an illegal procedure (number)
conf-factory@domain	Control instance of the conference bridges
conf@domain	Address of the conference bridge

4.2 Information and content

4.2.1 Outgoing communication

4.2.1.1 Communication request

Situation	The connected TE is in quiescent state according to 1TR110-1 [2], section 6.2.1. State P1 (quiescent)
Process within the TE	The TE goes off-hook (DC loop closed) - transition from quiescent state to loop state according to 1TR110-1 section 6.2.2.
Reaction by the VGW	The VGW shall ■ recognize the off-hook state, ■ initialize an instance for treating the address signal according to Annex A §F.1 „Sending of Invite with determining the end of address signalling / Annex A §F.1.1 Actions at the originating VGW", ■ play dial tone or special dial tone (according to the profile information) ■ provide a DTMF decoder (to receive dialling information), ■ provide optionally a decoder for loop disconnect dialling, ■ start timer T_{FD} (to monitor the end of acceptance of dialling signals). Transition to state P2 (steady loop state)
State a/b	Dial tone / special dial tone according to Annex C.

4.2.2 Communication set-up

Dialling information shall be handled according to ETSI TS 183 043, ANNEX F.1 „Sending of Invite with determining the end of address signalling".
In case of loop disconnect pulsing (LDP) the VGW shall interpret a pause greater or equal than the inter-digital pause (minimum pause between two LDP series) according to 1TR110-1 section 7.2.3 as the end of a pulse dialling digit. The VGW shall accept and process only digits which consist of 1 up to 10 loop pulses as correct dialling signals. Duration of greater than the end of dialling timer T_{a1} shall be interpreted as end of dialling.
The emergency numbers "110" and "112" shall be recognized by a Digit Map and shall be conveyed en-bloc in the INVITE message and sent out immediately.
The "#" sign (hash-key) shall be

- sent within the request SIP URI and the To header field if used as part of a service command code (SCC);
- discarded by the UE and not be sent out if used as "end of dialling" information.

4.2.2.1 Dialling

Situation	State P2 (steady loop state) Note: No dialling information has been received by the VGW, yet.
Process within the TE	The TE sends valid DTMF or LDP signals according to 1TR110-1 section 6.2.4 and section 7.
Reaction by the VGW	The VGW shall ■ detect the first valid dialling signal, ■ switch off the dial tone or special dial tone, ■ stop the timer T_{FD} , ■ start the timer T_{a1} ■ detect subsequent dialling signals ■ start the timer T_{a1} after each recognized dialling signal. Note: In case of LDP the VGW shall detect a pause of T_{IDP} as the end of a pulse dialling series. ($T_{IDP} = T_{a1}$)
State a/b	State P2 (steady loop state)

4.2.2.2 End of dialling

Situation	State P2 (steady loop state) and ■ Timer Ta1 [1] has expired or ■ a “#” sign has been recognized as end of dialling information or ■ the received dialling information have been interpreted as a routable address by means of a Digit Map.
Reaction by the VGW	The VGW shall ■ send a SIP request “INVITE” with the accumulated dialling information (adrinfo) in the user part of the address as following: `INVITE sip:adrinfo@domain, user=phone` ■ deactivate the detection circuits for DTMF signals and loop disconnect pulsing (LDP); ■ stop all timers regarding the dialling process. Transition to state P2.1 (end of dialling).
State a/b	State P2 (steady loop state)

[1] The expiry of timer T_{a1} shall be ignored while the evaluation of dialling information by the Digit Map is running.

4.2.2.3 Successful communication request

Situation	State P2.1 (end of dialling) VGW has received a SIP response „18x“. A: SIP response = „180 Ringing“ B: SIP response = „180 Ringing“ or „183 Session Progress“ with a P-Early-Media header.
Reaction within the VGW	The VGW shall A: Play ringing tone and start timer T_{RT} B: Establish media path according to „em-param“ and route received audio data to the port.
State a/b	A: Ringing tone according to Annex C. B: Audio data (e.g. announcement) No further changes.

4.2.2.4 Unsuccessful communication request

Situation	State P2 (outgoing communication request) or P2.1 (end of dialling). VGW has received a SIP response „4xx“, „5xx“ or „6xx“ A: There is still another communication on hold. B: No further communication.
Reaction within the VGW	The VGW shall ■ play audible tone according to Table 4-5, ■ start timer T_{BT} . A: Transition to state P5.2 (release request 2) B: Transition to state P5.1 (release request 1)
State a/b	Busy tone or congestion tone according to Annex C.

4.2.2.5 Abort of a communication request

Situation	State P2.1 (end of dialling)
Process within the TE	The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.
Reaction within the VGW	The VGW shall ■ detect the on-hook state, ■ reset the port related instances, ■ terminate an already established dialog by sending a SIP request „CANCEL“. Transition to state P1 (quiescent).
State a/b	State P1 (quiescent)

4.2.2.6 Called party’s answer

Situation	State P2.1 (end of dialling) On a communication request (INVITE) the VGW has received a SIP response „200 OK“.
Reaction within the VGW	The VGW shall ■ switch off the ringing tone (in case of a successful communication request according to clause 4.2.2.3 with a response = „180 Ringing“ without P-Early-Media header), ■ stop the timer T_{RT}, ■ establish the media path according to the SDP parameters, ■ route the audio data belonging to the media session to the port. If the response contains an XML document with AOC information than these information shall be handled according to clause 4.2.4.1. Transition to state P4 (communication).
State a/b	State P4 (communication).

4.2.3 Incoming communication

4.2.3.1 Communication request

Situation	State P1 (quiescent) , VWG receives SIP request: `INVITE sip:pUserID@domain`
Reaction within the VGW	The VGW shall ■ verify if all necessary resources are available.
State a/b	No change.

4.2.3.2 Communication acknowledgement

Situation	Communication request according to clause 4.2.3.1, VGW is able to process an incoming communication request.
Reaction within the VGW	The VGW shall ■ send a SIP response „180 Ringing“, ■ switch on the ringing signal, ■ start timer T_{Ring}. Depending on the port profile and the requirements in Table 4-3 and Table 4-4, FSK data transmission shall be sent. Transition to state P3 (ringing state).
State a/b	Ringing signal (AC/DC conditions) according to Annex C. Depending on the port profile, FSK data transmission shall be sent between the first and second ringing pattern according to Annex C.

4.2.3.3 Communication waiting request

Situation	State P4 (communication), VGW receives SIP request: INVITE sip:pUserID@domain An XML document containing a Communication Waiting Indication can be provided, additionally.
Reaction within the VGW	The VGW shall ■ verify if all necessary resources are available.
State a/b	No change.

4.2.3.4 Communication waiting acknowledgement

Situation	Communication waiting request according to clause 4.2.3.3. VGW is able to process the request.
Reaction within the VGW	The VGW shall ■ send a SIP response „180Ringing“. Depending on the port profile and the requirements in Table 4-3 and Table 4-4, FSK data transmission and the CW tone shall be sent via the relevant port. During the FSK data transmission the voice path to and from the far-end party shall be cut off.
State a/b	The communication between subscriber A and B is unchanged, except during FSK data transmission. CW tone and depending on the port profile, FSK data transmission shall be sent according to Annex C.

4.2.3.5 Abort of the communication waiting request

Situation	Communication waiting request according to clause 4.2.3.4. The waiting party (subscriber C) has gone on-hook or the no answer timer has expired. VGW receives SIP request: CANCEL sip:pUserID@domain
Reaction within the VGW	The VGW shall ■ switch off the CW tone, ■ reset the port related instances concerning the waiting communication.
State a/b	The communication between subscriber A and B is unchanged. CW tone aborted.

4.2.3.6 Contrary communication request

Situation	State P2 (steady loop state), P2.1 (end of dialling) or P5.1 to P5.3 (release request 1 to 3). VGW receives SIP request: `INVITE sip:pUserID@domain`
Reaction within the VGW	The VGW shall ■ reject the incoming communication request using the SIP response „486 Busy here“.
State a/b	No change.

4.2.3.7 Abort of the communication request

Situation	State P3 (incoming communication) The calling party has gone on-hook or the communication request has been aborted by the Call Control due to timeout. VGW receives SIP request: `CANCEL sip:pUserID@domain`
Reaction within the VGW	The VGW shall ■ switch off the ringing signal, ■ stop timer T_{Ring}. Transition to state P1 (quiescent).
State a/b	State P1 (quiescent)

4.2.3.8 Called party's answer

Situation	State P3 (incoming communication; ringing state).
Process within the TE	The TE goes off-hook (DC loop closed) - transition from ringing state to communication state according to 1TR110-1 section 6.2.8.
Reaction within the VGW	The VGW shall ■ detect the off-hook state, ■ send a SIP response „200 OK“, ■ switch off the ringing signal, ■ stop the timer T_{Ring}, ■ establish the media session, ■ provide DC potential. Transition to state P4 (communication).
State a/b	State P4 (communication).

4.2.4 Advice of charge (AOC)

Advice of charge (AOC) information shall be treated according to ETSI TS 124 447 [6], Annex D „AOC XML Schema".

During a communication different AOC types (e.g. AOC-D) can be used to convey tariff information. All types of AOC described in [6] are supported, depending on the requested service options on the subscriber line.

NOTE: For analogue ports the IAD acts as Charging Generation Point.

AOC tariff information will only be conveyed in monetary format (currency), but not in non-monetary format (units).
If two communications have been established via the VGW (e.g. in case of HOLD, Toggle or 3PTY), the VGW receives separate AOC XML's for each communication.

All AOC information received during a communication can be mapped to 16-kHz metering pulses and provided on the relevant port. In case of block tariff any new tariff information (e.g. due to tariff modification) shall not disrupt the running transmission of metering pulses.
A premature stop of sending metering pulses may occur if:

- an on-hook condition has been recognized at the relevant port
- a "BYE" request has been received (in this case a running metering pulse or a block of metering pulses shall be completely transmitted).

NOTE: The mapping from SIP AOC information to 16kHz metering pulses is described in [8].

4.2.4.1 Advice of charge at communication set-up (AOC-S)

The AOC-S service provides tariff information at the beginning of a communication and when tariff changes occur during a communication.
AOC-S information can be conveyed in the following SIP messages:

- reliable 18x message
- 200OK (INVITE)
- INFO

The AOC-S tariff-information can be expressed as:

- Monetary price per time unit and time unit (basic-type->price-time);
- Monetary flat rate (communication-setup->flat-rate).

The VGW stores the monetary AOC-S charging rate and calculates the timing for 16-kHz pulses based on the AOC-S charging rate and the provisioned monetary price per 16-kHz pulse in the VGW.

NOTE: For a reliable presentation of the charging rate (monetary format) on an analogue TE, the same price per 16-kHz unit needs to be set in the TE.

Situation	Transition to state P4 (communication) VGW has received a SIP response (e.g. „200 OK") with an

	XML document including the following AOC information: charged-itemsType = basic
Reaction within the VGW	The VGW should ■ calculate and map these AOC tariff information into 16-kHz metering pulses.
State a/b	State P4 (communication) ; 16-kHz metering pulses according to Annex C.

4.2.4.2 Advice of charge during communication (AOC-D)

The AOC-D service provides tariff information about the recorded charges during the active phase of a communication.
Only accumulated charges in monetary format are transmitted; therefore only the first AOC-D tariff information and afterwards the difference between the last and the latest tariff information is used to calculated the timing for 16-kHz pulses.
AOC-D information can be conveyed in the following SIP messages:

- INFO
- BYE
- Final response (BYE)

The VGW stores the monetary AOC-D charging information and calculates the timing for 16-kHz pulses based on the provisioned monetary price per 16-kHz pulse in the VGW.

NOTE: For a reliable presentation of the charging rate (monetary format) on an analogue TE, the same price per 16-kHz unit needs to be set in the TE.

Situation	State P4 (communication) VGW has received a SIP request „INFO“ with an XML document including the following AOC information: ■ type of charging information = total ■ recorded-charges
Reaction within the VGW	The VGW should ■ calculate and map these AOC tariff information into 16-kHz metering pulses.
State a/b	State P4 (communication) ; 16-kHz metering pulses according to Annex C.

4.2.4.3 Advice of charge during communication (AOC-E)

The AOC-E service provides tariff information about the recorded charges for a communication when it is terminated.
Only accumulated charges in monetary format are transmitted; those need not to be converted to 16-kHz pulses.

NOTE: AOC-E should not be chosen for analogue ports, since the conversion to 16-kHz pulses and their recognition within the TE after transition to the on-hook state is not ensured.

4.2.5 End of communication

4.2.5.1 Communication released by party A

Situation	State P4 (communication)
Process within the TE	The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.
Reaction within the VGW	The VGW shall ■ detect the on-hook state, ■ send a SIP request: `BYE sip: cc@domain` ■ disconnect the related media session, ■ reset all port related instances. Transition to state P1 (quiescent).
State a/b	State P1 (quiescent)

4.2.5.2 Communication released by party B

Situation	State P4 (communication) VGW receives SIP request: `BYE sip:pUserID@domain`
Reaction within the VGW	The VGW shall ■ send a SIP response “200 OK”. Release request according to clause 4.2.9.7.
State a/b	No change.

4.2.6 Communication waiting (CW)

4.2.6.1 CW acceptance by hang-up (re-ringing)

Situation	State P4 (communication between party A and B) and incoming communication request (party C) for party A according to clause 4.2.3.3; CW tone is played and depending on the user profile, FSK data transmission (e.g. CLIP) shall take place; or state P4.1 (Hold 1).
Process within the TE	The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.
Reaction within the VGW	The VGW shall ■ detect the on-hook state, ■ send a SIP request “BYE” (concerning the communication between A and B), ■ switch on the ringing signal,

- start timer T_{RRing} .

	Transition to state P3 (ringing state) regarding the new communication between party A and C.
State a/b	Ringing signal according to Annex C. Depending on the port profile, FSK data transmission shall be sent between the first and second ringing pattern according to Annex C.

4.2.6.2 Abort of the re-ringing

Situation	State P3 (incoming communication) related to the communication between A and C and a SIP request "CANCEL" has been received (supervisory timer has expired in the network).
Reaction within the VGW	The VGW shall ■ switch off the ringing signal. Transition to state P1 (quiescent).
State a/b	State P1 (quiescent)

4.2.6.3 CW handling using service procedures

Situation	State P4 (communication between A and B); Incoming communication request (party C) for party A according to clause 4.2.3.3; CW tone is played and depending on the user profile, FSK data transmission (e.g. CLIP) takes place.
Process within the TE	TE performs a hook-flash signal (initiated by subscriber A) according 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ recognize the hook-flash signal, ■ deactivate the current media session and put party B on hold, ■ switch off the CW tone and play the special dial tone, ■ provide a DTMF decoder (to receive the SOC), ■ start timer T_{FD}. Transition to state P4.3 (service) related to the communication between party A and B. During state P4.3, this media session has to be kept deactivated.
State a/b	Special dial tone according to Annex C.

4.2.6.4 Switching Order Command „0" (SOC)

Situation	State P4.3 (service) related to the communication between party A and B. (Target: Return to current communication between party A and B; rejection of the communication of party C).
Process within the TE	TE sends DTMF signal „0".
Reaction within the VGW	The VGW shall

- switch off the special dial tone,
- stop the timer T_{FD} ,
- send a SIP response „603 Decline"(reject the waiting communication),
- activate the previous media session, again,
- return to state P4 (communication).

State a/b — State P4 (communication).

4.2.6.5 Switching Order Command „1"

Situation	State P4.3 (service) related to the communication between party A and B. (Target: Release current communication between party A and B; activate the waiting communication between party A and C).
Process within the TE	TE sends DTMF „1".
Reaction within the VGW	The VGW shall

- switch off the special dial tone,
- stop the timer T_{FD} ,
- send a SIP request "BYE" (release previous communication),
- send a SIP response „200 OK"(accept the waiting communication),
- activate the new media session between party A and C.

Transition to state P1 (quiescent) concerning the previous communication between party A and B. [2]

State a/b — State P4 (communication).

4.2.6.6 Switching Order Command „2"

Situation	State P4.3 (service) related to the communication between party A and B. (Target: Keep current communication with party B on hold; activate the waiting communication between party A and C).
Process within the TE	TE sends DTMF „2".
Reaction within the VGW	The VGW shall

- detect the DTMF signal "2",
- switch off the special dial tone,
- stop the timer T_{FD},
- send a SIP response „200 OK"(accept the waiting communication),
- send a SIP request "re-INVITE" with the SDP media

[2] The port states of the communication between party A and C (P4: communication) are out of the scope of the present document (see clause 4.1).

attribute: a=sendonly,

- activate the new media session between party A and C.

	Transition to state P4.1 (hold 1) concerning the previous communication with party B. [2]
State a/b	State P4 (communication).

4.2.7 Hold/Call hold (HOLD)

4.2.7.1 Invoke HOLD

Situation	State P4 (communication)
Process within the TE	TE performs a hook-flash signal (initiated by subscriber A) according 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ recognize the hook-flash signal, ■ send a SIP request “re-INVITE” with the SDP media attribute: a=sendonly, ■ deactivate the current media session and put party B on hold, ■ play the special dial tone, (after SIP response “200 OK” has been received), ■ provide a DTMF decoder (to receive dialling information), ■ start timers T_{FD} and T_{a1} (see clause 4.2.2). Transition to state P4.1 (Hold 1) related to the communication between party A and B.
State a/b	Special dial tone according to Annex C.

4.2.7.2 Call hold

Situation	State P4.1 (Hold 1).
Process within the TE	TE sends dialling information.
Reaction within the VGW	The VGW shall ■ correctly receive and evaluate the dialling information. The communication set-up to party C proceeds according to clause 4.2.2. [3] After a successful communication set-up between party A and C, the state concerning the communication between party A and B change to state P4.2 (Hold 2). AOC information during this communication shall be treated according to clause 4.2.4.
State a/b	State P4.2 (Hold 2)

[3] The port states of the communication set-up to party C as well as the communication between party A and C (P4: communication) are out of the scope of the present document (see clause 4.1).

4.2.7.3 End of HOLD state in case of an invalid input

Situation	State P4.1 related to the communication with party B (Hold 1). The communication set-up (call hold) has been aborted according to clause 4.2.9.3. Transition to state P4.3 (service).
Process within the TE	TE sends DTMF “1”.
Reaction within the VGW	The VGW shall ▪ switch off the special dial tone, ▪ stop the timer T_{FD} , ▪ send a SIP request “re-INVITE” with the SDP media attribute: a=sendrecv, ▪ activate the previous media session between party A and B. Transition to state P4 (communication) regarding the communication between party A and B.
State a/b	State P4 (communication).

4.2.7.4 End of HOLD state due to hang-up (re-ringing)

Situation	State P4.1 related to the communication with party B (Hold 1) or P4.2 (Hold 2).
Process within the TE	The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.
Reaction within the VGW	The VGW shall ▪ detect the on-hook state, ▪ send a SIP request “BYE” (if a call back communication has been successfully established before: state P4.2), ▪ switch on the ringing signal, ▪ start timer T_{RRing} . Transition to state P3 (incoming communication) regarding the previous communication between party A and B. [4].
State a/b	Ringing signal according to Annex C. Depending on the port profile, FSK data transmission shall be sent between the first and second ringing pattern according to Annex C.

[4] The port states of the call back communication between party A and C are out of the scope of the present document (see clause 4.1).
Only the re-ringing of the previous communication with party B is considered.

4.2.7.5 End of re-ringing (timeout)

Situation	State P3 (ringing state); timer T_{RRing} has expired.
Reaction within the VGW	The VGW shall ■ switch off the ringing signal, ■ send a SIP request “BYE” (reject the incoming call). Transition to state P1 (quiescent).
State a/b	State P1 (quiescent).

4.2.7.6 Invoke the service procedure

Situation	State P4.2 (Hold 2) related to the communication between party A and B. The communication (state P4) between party A and C is active.
Process within the TE	TE sends a hook-flash according to 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ detect the hook-flash signal, ■ interrupt the current media path between party A and C, ■ play the special dial tone, ■ provide a DTMF decoder (to receive the SOC), ■ start timer T_{FD} , Transition to state P4.3 (service) related to the communication between party A and B. During state P4.3, this media session has to be kept deactivated. [5]
State a/b	State P4.3 (service); Special dial tone according to Annex C.

4.2.7.7 SOC “0”

Situation	State P4.3 (service) related to the communication between party A and B. The communication between party A and C is active. (Target: Release the communication with party B on hold; return to the previously active communication between party A and C).
Process within the TE	TE sends DTMF “0”.
Reaction within the VGW	VWG shall ■ detect the DTMF signal „0“, ■ switch off the special dial tone, ■ stop the timer T_{FD} , ■ send a SIP request “BYE” (concerning the communication with party B on hold) ■ activate the previous media session between party A

[5] The port states of the communication between party A and C (P4: communication) are out of the scope of the present document (see clause 4.1).

and C.

Transition to state P1 (quiescent) regarding the communication with party B. [5]

State a/b	State P4 (communication).

4.2.7.8 SOC "1"

Situation	State P4.3 (service) related to the communication between party A and B. The communication between party A and C is active. (Target: Release the previous communication with party C on hold; return to the inactive communication between party A and B). [6]
Process within the TE	TE sends DTMF "1".
Reaction within the VGW	The VGW shall

- detect the DTMF signal „1",
- switch off the special dial tone,
- stop the timer T_{FD} ,
- send a SIP request "BYE" (concerning the communication with party C on hold),
- wait until 200 OK has been received,
- send a SIP request "re-INVITE" with the SDP media attribute: a=sendrecv (concerning the communication with party B on hold),
- activate the previously inactive media session between party A and B.

Transition to state P4 (communication) regarding the communication between party A and B.

State a/b	State P4 (communication).

4.2.7.9 SOC "2"

Situation	State P4.3 (service) related to the communication between party A and B. The communication between party A and C is active. (Target: Put the previously active communication with party C on hold and activate the previously inactive communication with party B – toggle function). [6]
Process within the TE	TE sends DTMF "2".
Reaction within the VGW	The VGW shall

- detect the DTMF signal „2",
- switch off the special dial tone,
- stop the timer T_{FD} ,
- send a SIP request "re-INVITE" with the SDP media attribute: a=sendonly (concerning the previously active

[6] The information regarding the communications refers to the first transition to state P4.1 (service) as described in 4.2.7.6, only.

communication),

- send a SIP request “re-INVITE” with the SDP media attribute: a=sendrecv (concerning the previously inactive communication),
- activate the previously inactive media session between party A and B.

Transition to state P4 (communication) regarding the previously inactive communication between party A and B.

State a/b — State P4 (communication).

4.2.8 Three-Party-Conference (3PTY/CONF)

The Service is currently not supported as an AS feature, therefore everything that belongs to AS procedures are not valid and has to be fulfilled within the VGW.

4.2.8.1 Invoke the service procedure

Situation — State P4.2 (Hold 2) related to the communication between party A and B.
The communication between party A and C is active.

Process within the TE — TE sends hook-flash according to 1TR110-1 section 6.2.5.3.

Reaction within the VGW — The VGW shall

- detect the hook-flash signal,
- interrupt (mute) the current media path between party A and C, (Only locally in VGW, no SIP actions towards user C)
- play the special dial tone,
- provide a DTMF decoder (to receive the SOC),
- start timer T_{FD} ,

Transition to state P4.3 (service) related to the communication between party A and B. During state P4.3, this media session has to be kept deactivated.

State a/b — State P4.3 (service);
Special dial tone according to Annex C.

4.2.8.2 SOC “3”

Situation — State P4.3 (service) related to the communication between party A and B.
The communication between party A and C is active.
(Target: Invoke a conference between party A, B and C).

Process within the TE — TE sends DTMF “3”.

Reaction within the VGW — The VGW shall

- detect the DTMF signal „3“,
- switch off the special dial tone,
- stop the timer T_{FD} ,
- Built CONF Focus (Context) where all 3 RTP (A, B and C-Party) shall be mixed within the VGW.

- send a SIP request “re-INVITE” with the SDP media attribute: a=send sendrecv (concerning the previously held communication),
- activate the " CONF focus" within the VGW of the served user
- activate (unmute) the current media path between party A and C, (Only locally in VGW, no SIP actions towards user C since A-C is still in “sendrecv”)

Transition to state P4.4 (conference) when the SIP response “200 OK” has been received.

State a/b	State P4.4 (conference); Communication (three party conference).

4.2.8.3 Invoke the service procedure (return to call hold)

Situation	State P4.4 (conference)
Process within the TE	TE sends hook-flash signal according to 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall

- detect the hook-flash signal,
- interrupt (mute) the current media path between party A and C, as well as party A and B (Only locally in VGW, no SIP actions towards user C)
- play the special dial tone towards Party A,
- provide a DTMF decoder (to receive the SOC),
- start timer T_{FD} .

Transition to state P4.3 (service).

State a/b	State P4.3a (SOC interaction); Special dial tone according to Annex C.

4.2.8.4 SOC “2”

Situation	State P4.3a (SOC interaction); (Target: Stop the three party conference state. The active communication before the conference shall be re-established between party A and C; the communication with party B was on hold).
Process within the TE	TE sends DTMF “2”.
Reaction within the VGW	The VGW shall

- detect the DTMF signal „2“,
- switch off the special dial tone,
- stop the timer T_{FD} ,
-
- send a SIP request “re-INVITE” with the SDP media attribute: a=sendonly to Party B

(to put the last inactive communication before the 3PTY conference has been invoked on hold),

- send a SIP request "re-INVITE" with the SDP media attribute: a=sendrecv to Party C (normally not needed if credentials sent within the SIP re-INVITE does not change)

- Activate (unmute) the current media path between party A and C internally the VGW if needed.

Transition to state P4.2 (Hold 2) concerning the communication between party A and B (party B on hold and communication between party A and C active).

State a/b — State P4 (communication).

4.2.8.5 End of 3PTY conference due to hang-up by the initiator

Situation — State P4.4 (conference)

Process within the TE — The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.

Reaction within the VGW — The VGW shall

- detect the on-hook state,
- send a SIP request "BYE" (concerning the communication with party B)
- send a SIP request "BYE" (concerning the communication with party C)
- terminate internal 3PTY conference bridge (Disconnect the related media sessions and reset all port related intances).
- SIP response "200 OK" for BYE is received.

Transition to state P1 (quiescent).

State a/b — State P1 (quiescent).

4.2.8.6 End of 3PTY conference due to hang-up by party B

Situation — State P4.4 (conference)
VGW receives a SIP request "BYE" from party B.

Reaction within the VGW — The VGW shall

- Send SIP response "200 OK" for BYE.
- send a SIP request "re-INVITE" with the SDP media attribute: a=sendrecv to activate the communication between party A and C,
-

Transition to state P4 (communication) [7]

State a/b — State P4 (communication).

---

[7] In case of hang-up by party B a transition to state P1 (quiescent) takes place.

4.2.9 Special procedures

4.2.9.1 Rejection of a communication request

Situation	The VGW is not able to process an incoming communication request according to clause 4.2.3.1 or 4.2.3.3.
Reaction within the VGW	The VGW shall ■ send a SIP response "500 internal server error".
State a/b	No change.

4.2.9.2 Dialling of unallowable digits

Situation	State P4.3 (service)
Process within the TE	TE sends an unallowable digit or character (DTFM or PD signal) depending on the service feature.
Reaction within the VGW	The VGW shall ■ stop all timers regarding the dialling state, An active communication shall be put on hold by sending a SIP request „re-INVITE" with the SDP media attribute: a=sendonly. Internal announcement should be played (e.g. „Dienst oder Dienstmerkmal nicht möglich").
State a/b	Announcement (e.g. „Dienst oder Dienstmerkmal nicht möglich").

4.2.9.3 Quit the error state by hook-flash

Situation	State P4.3 (service); Dialling of an unallowable digit according to 4.2.9.2
Process within the TE	TE performs another hook-flash signal (e.g. initiated by the subscriber to quit the error state) according to 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ detect the hook-flash signal, ■ ■ play the special dial tone, ■ provide a DTMF decoder (to receive the SOC), ■ start timer T_{FD} , State P4.3 (service) is kept.
State a/b	State P4.3 (service); Special dial tone according to Annex C.

4.2.9.4 Quit the error state by hang-up

Situation	State P4.3 (service); Dialling of an unallowable digit according to 0
Process within the TE	The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.
Reaction within the VGW	The VGW shall ■ detect the on-hook state, ■ establish the ringing signal on the addressed port ■ start timer T_{RRing} . Transition to state P3 (ringing state); Depending on the port profile and the requirements in Table 4-3 and Table 4-4, FSK data transmission shall be sent. If the recall is not answered by a transition to state P4 (communication) within the timeout of T_{RRing} the hold party shall be released according to 4.2.7.5.
State a/b	Ringing signal according to Annex C. Depending on the port profile, FSK data transmission shall be sent between the first and second ringing pattern according to Annex C.

4.2.9.5 Unexpected hook-flash

Situation	State P4.1 (Hold 1), P4.2 (Hold 2) or P4.3 (Service).[8] VGW expects DTMF signals (dialling digits or SOC).
Process within the TE	TE performs an unexpected hook-flash signal according to 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ detect the hook-flash signal, ■ stop all timers regarding the dialling state, ■ play the congestion tone.
State a/b	Congestion tone according to Annex C.
Process within the TE	TE performs another hook-flash signal (e.g. initiated by the subscriber to quit the error state) according to 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ detect the hook-flash signal, ■ switch off the congestion tone, ■ play the special dial tone, ■ provide a DTMF decoder (to receive the SOC), ■ start timer T_{FD} . Transition to state P4.3 (service).
State a/b	Special dial tone according to Annex C.

[8] A hook-flash shall be ignored during the states P2, P2.1, P5.1 or P5.3.

4.2.9.6 Hook-flash during release request

Situation	State P5.2 (release request 2).
Process within the TE	TE performs a hook-flash signal according to 1TR110-1 section 6.2.5.3.
Reaction within the VGW	The VGW shall ■ detect the hook-flash signal, ■ stop the timer T_{BT} , ■ play the special dial tone, ■ provide a DTMF decoder (to receive the SOC), ■ start timer T_{FD} , Transition to state P4.3 (service).
State a/b	Special dial tone according to Annex C.

4.2.9.7 Transition to release request

Situation	One of the following causes has taken place: ■ timer T_{FD}, T_{BT} or T_{RT} has expired, ■ an active communication has been released by party B (see 4.2.5.2). A: A second communication is on hold (party C), B: No further communication is involved, C: Timer T_{BT} has expired in state P5.2.
Reaction within the VGW	The VGW shall ■ play the congestion tone, ■ start timer T_{TB} . Case A: An active dialogue shall be terminated; Transition to state P5.2 (release request 2). Case B and C: Terminate all active dialogues; Reset all port related instances; Transition to state P5.3 (release request 3: ULS state).
State a/b	Congestion tone according to Annex C.

4.2.9.8 Supervision of release request

Situation
A: State P5.1 or P5.3 (release request 1 or 3);
B: State P5.2 (release request 2).
The TE goes on-hook (DC loop open) – transition to the quiescent state according to 1TR110-1 section 6.2.1.

Reaction within the VGW
The VGW shall

- detect the on-hook state,
- stop the timer T_{BT} (if necessary),

Case A:

Transition to state P1 (quiescent).

Case B:

- establish the ringing signal on the addressed port,
- start timer T_{RRing} ,

Transition to state P3 concerning the waiting communication (incoming communication request).

State a/b
A: Quiescent state according to 1TR110-1 section 6.2.1.
B: Ringing signal according to Annex C.

4.2.9.9 Communication request during release request

Situation
State P5.1 until P5.3 (release request 1 to 3),
VGW receives SIP request:

```
INVITE sip:pUserID@domain
```

Reaction within the VGW
The VGW shall

- reject the incoming communication request with SIP response “486 Busy here”.

State a/b
No change.

4.2.9.10 Timeout of an incoming communication request

Situation
State P3 (ringing state).
Timer T_{Ring} has expired.

Reaction within the VGW
The VGW shall

- switch off the ringing signal (at the relevant port),
- send a SIP request “480 Temporarily unavailable” (reject the incoming call).

Transition to state P1 (quiescent).

State a/b
State P1 (quiescent).

4.3 Tables

#	State	Remark
P1	Quiescent state	Analogue port is ready (TE in idle state; floating potential between a- and b-wires).
P2	Line seizure	Starts after transition from quiescent state to steady loop state (line seizure: TE has changed to off-hook state) until beginning of dialling.
P2.1	End of dialling	An INVITE request with address information has been sent out.
P3	Incoming communication request (ringing state)	Analogue port is ready and ringing resources are available. A SIP INVITE request has been received.
P4	Communication	An incoming or outgoing communication request has been acknowledged with „200 OK“.
P4.1	Hold 1	After recognition of a hook-flash, the active communication is put on hold. The special dial tone shall be played. Waiting for dialling information.
P4.2	Hold 2	A second active communication (e.g. A-C) has been established while the first communication (e.g. party B) is on hold.
P4.3	Service	The first communication (e.g. party B) is still on hold. After recognition of a hook-flash, the previously active communication (e.g. party C) is deactivated. The special dial tone shall be played. Waiting for receipt of a SOC.
P4.3a	SOC interaction	At the VGW, two active communications still remain in the three party conference. The media path between party A- and –B as well as party A- and –C are interrupted (mute). The special dial tone shall be played to the initiator of the conference. Waiting for receipt of a SOC.
P4.4	Conference	On an analogue port, two active communications have been interconnected to a three party conference.
P5.1	Release request 1	A communication set-up was unsuccessful according to 4.2.2.4. Busy tone or congestion tone according to Annex C shall be played. No further communication is on hold.
P5.2	Release request 2	A communication set-up was unsuccessful according to 4.2.2.4 or another event according to 4.2.9.7 takes place which terminates the communication. No further communication is on hold.
P5.3	Release request 3	An event according to 4.2.9.7 takes place which terminates

		the communication.

Table 4-1 „Port states of the VGW“

Timer	Time-out	Start	Stop	Action after timeout
T_{FD}	60 ± 2 s (according to 1TR110-1)	Line seizure request according to clause 4.2.1.1.	Receipt of first dialling information according to clause 4.2.2.1.	Release request according to clause 4.2.9.7.
T_{1a}	4 s to 12 s (according to Annex A)	At receipt of fresh dialling information according to clause 4.2.2.1.	At receipt of fresh dialling information according to clause 4.2.2.1 or at receipt of a “#” (hash) sign (end of dialling information) or the dialling information has been recognized by Digit Map as a routable address.	Disable DTMF/LDP receiver; Send INVITE; End of dialling according to clause 4.2.2.2.
T_{RRing}	15 ± 1 s	A communication is on hold according to 4.2.7.4; TE goes on-hook (DC loop open).	TE goes off-hook (DC loop closed).	Stop re-ringing.
T_{Ring}	180 ± 2 s	Seizure acknowledgement on an incoming communication according to 0.	TE goes off-hook (DC loop closed).	Release ringing signal.
T_{BT}	60 ± 2 s	Unsuccessful communication set-up according clause 4.2.2.4 or release request initiated by subscriber B.	TE goes on-hook (DC loop open).	Release request according to clause 4.2.9.7.
T_{RT}	180 ± 2 s	On receipt of „180 Ringing“ ringing tone applies.	The connection has been released by subscriber A or B.	Release request according to clause 4.2.9.7.

Table 4-2 „Timer“

INVITE		Interface (a/b port)
From Header	**History Header with cause value according to RFC 4458, §2.2**	**FSK data transmission according to 1TR110-1 Anhang B**
unsubscribed@...	arbitrary	No FSK data transmission
anonymous@...	arbitrary	Reason for Absence of CLI parameter: "CLIR" (04h); Parameter Content: "Private (CLIR involved)" (50h)
unavailable@...	arbitrary	Reason for Absence of CLI parameter: "CLIR" (04h); Parameter Content: "Private (Unavailable)" (4Fh)
CgPN@...[9]	not available	Call Setup message parameter: "Calling Line Identity" (02h); Digits: from the From:- Header
	available	Call Setup message parameter: "Calling Line Identity" (02h); Digits: from the From:- Header Type of Forwarded Call: "Forwarded connection" (15h); Parameter Content: "Unavailable or unknown forwarded call type" (00h)

Table 4-3 „Mapping SIP header parameters to FSK information" [10]

INVITE		Interface (a/b port)
From Header	**CCBS/CCNR/CCNL indication**	**FSK data transmission according to 1TR110-1 Anhang B**
unsubcsribed@...	arbitrary	No FSK data transmission
CgPN@...[9]	available	Call Setup message parameter: "Calling Line Identity" (02h), Digits: from the From:- Header Call Type parameter: "CCBS callback" (11h), Parameter Content: "CCBS/CCNR" (02h)

Table 4-4 „Mapping SIP header parameters to FSK information" [10]

[9] If the CgPN includes a "+" sign, it shall be mapped to "00" in the CLI parameter.
[10] The Privacy Header shall not be evaluated.

Cause	Audible tone (according to Annex C)
Off-hook; transition to state P2 (outgoing line seizure); as prescribed in user profile: name= „dial-tone-pattern“ (default = „standard-dial-tone“.	Dial tone
Like „Dial tone“; as prescribed in user profile: name= „dial-tone-pattern“, value = „special-condition-tone“.	Special dial tone
SIP INVITE request with CW indication during state P4 (communication)	Communication waiting tone
Receipt of a SIP response “180 Ringing” without P-Early-Media header.	Ringing tone
Receipt of a SIP response “486 Busy Here”	Busy tone
After receipt of any 4xx, 5xx or 6xx responses as well as on receipt of a BYE or CANCEL request in state P4 (communication).	Congestion tone

Responses with Reason-Header, cause = [11]		Audible tone
16	„Normal call clearing“	Congestion tone
17	„User busy“	Busy tone
18	„No user responding“	Busy tone
19	„No answer from user (user alerted)”	Busy tone
28	„Address incomplete“	Congestion tone
31	„Normal, unspecified“	Congestion tone
34	„No circuit/ channel available“	Congestion tone
38	„Network out of order“	Congestion tone
41	„Temporary failure“	Congestion tone
44	„Requested circuit/channel not available”	Congestion tone
47	„Resources unavailable, unspecified”	Congestion tone
58	„Bearer capability not implemented“	Congestion tone
91	„Invalid transit selection“	Congestion tone
102	„Recovery on timer expiry“	Congestion tone
127	„Interworking unspecified“	Congestion tone

Table 4-5 „Audible tones and their relevant causes“

[11] Causes according to Q.850. On deviant cause values the “Congestion tone” shall be played.
If the Reason header is missing, the audible tone regarding the SIP response/request (upper part of Table 4-5) shall be played.

<table>
<tr><th rowspan="2">Cause</th><th colspan="2">Port state</th></tr>
<tr><th>P5.1</th><th>P5.3</th></tr>
<tr><td>No dialling, T_{FD} has expired.</td><td rowspan="4">Transition to P5.3, immediately.</td><td rowspan="6">Congestion tone (60s),
reset after on-hook, only.
State: Unnecessary line seizure (ULS).</td></tr>
<tr><td>Communication request at party B.
Ringing tone has expired (120s).</td></tr>
<tr><td>Communication has been released by party B.</td></tr>
<tr><td>Error (e.g. no resources).</td></tr>
<tr><td>Unsuccessful connection (4xx, 5xx, 6xx).</td><td>Congestion tone (60s),
reset after on-hook, only.</td></tr>
<tr><td>Subscriber is busy (486 ...).</td><td>Busy tone (60s),
reset after on-hook, only.</td></tr>
</table>

Table 4-6 „Transitions into unnecessary line seizure state (ULS)" [12]

<table>
<tr><th rowspan="2">Cause</th><th colspan="2">Port state</th></tr>
<tr><th>P5.2</th><th>P5.3</th></tr>
<tr><td>No dialling, T_{FD} has expired.</td><td rowspan="8">Congestion tone (60s),
Hook-flash
or
reset after on-hook.</td><td rowspan="9">Congestion tone (60s),
reset after on-hook, only.
State: Unnecessary line seizure (ULS).</td></tr>
<tr><td>Communication request at party B.
Ringing tone has expired (120s).</td></tr>
<tr><td>Communication has been released by party B.</td></tr>
<tr><td>Error (e.g. no resources).</td></tr>
<tr><td>Communication error (e.g. wrong SOC).</td></tr>
<tr><td>Hook-flash during dialling state.</td></tr>
<tr><td>Hook-flash during service state.</td></tr>
<tr><td>Unsuccessful connection (4xx, 5xx, 6xx).</td></tr>
<tr><td>Subscriber is busy (486 ...).</td><td>Busy tone (60s),
Hook-flash or reset after on-hook.</td></tr>
</table>

Table 4-7 „Transitions into unnecessary line seizure state (ULS) during HOLD or 3PTY" [13]

[12] These transitions refer to an active communication (for information only).
[13] Besides an active communication, a second communication is on hold on the same port (for information only).

4.4 Modifications to ETSI TS 183 043

The relevant modifications in ETSI TS 183 043 for SIP UE (e.g. IAD) intended to be connected to the NGN platform of Deutsche Telekom are provided in Annex A of the present document.

Annex A ETSI TS 183 043 V2.3.1 (2009 03); 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 !

The protocol specification for IMS-based PSTN/ISDN emulation for VGW (IAD) intended to be connected to xDSL accesses of Deutsche Telekom are described in the specifically modified ETSI Standard TS 183 043 V2.3.1 (2009 03). The modifications in this ETSI standard assure the compatibility with the NGN platform of Deutsche Telekom.

The modified specification is available as a PDF file with the following file name:

- **1TR126 Annex A_V110_TS183043v020301p.pdf**

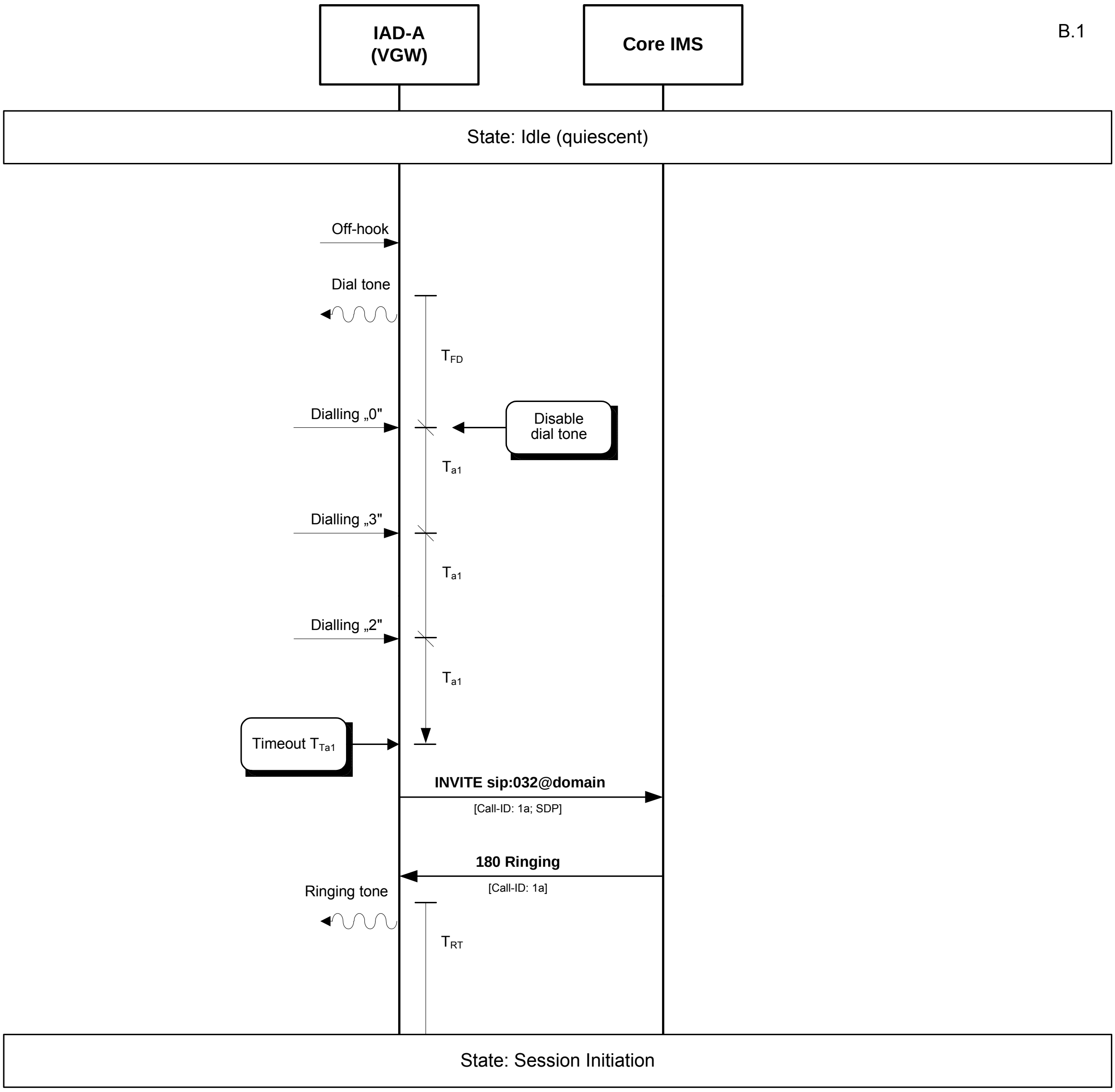
NOTE: The modified text that is added or deleted is shown as cursive and underlined (*<u>example for added text</u>*) or cursive and stricken (*~~example for stricken text~~*).

Annex B: Call flows for UE (IAD) supporting analogue interfaces (informative)

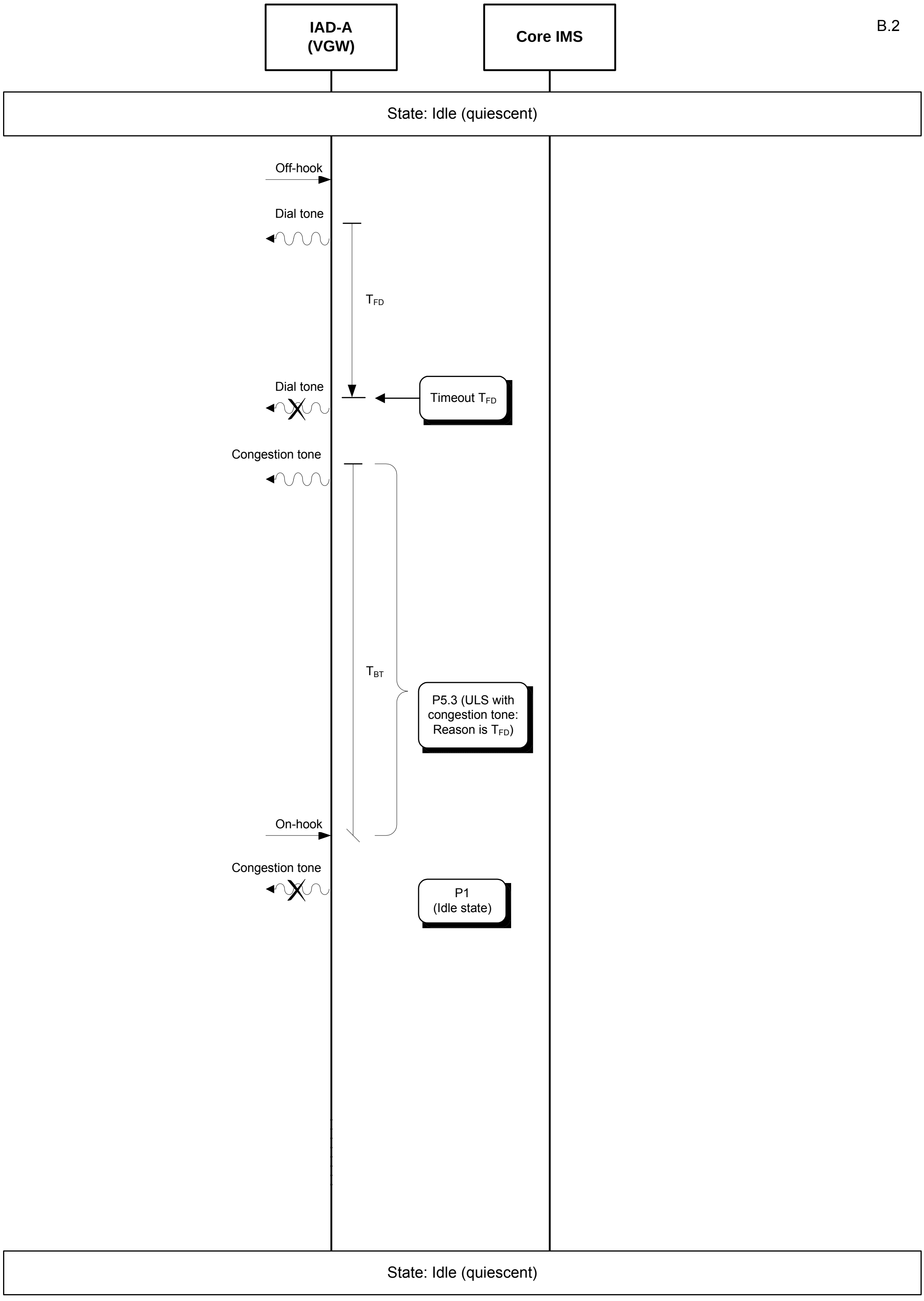
This annex B contains the call flows between the analogue interface and the SIP (Gm) interface. The SIP procedures are general valid also for pure SIP UE.

Call Flows were updated due to the fact that Call Wait, Call HOLD, 3PTY and TOGGLE shall be provided by the end device. Announcement that user is held will not be played by the network. This can be done by the end device.

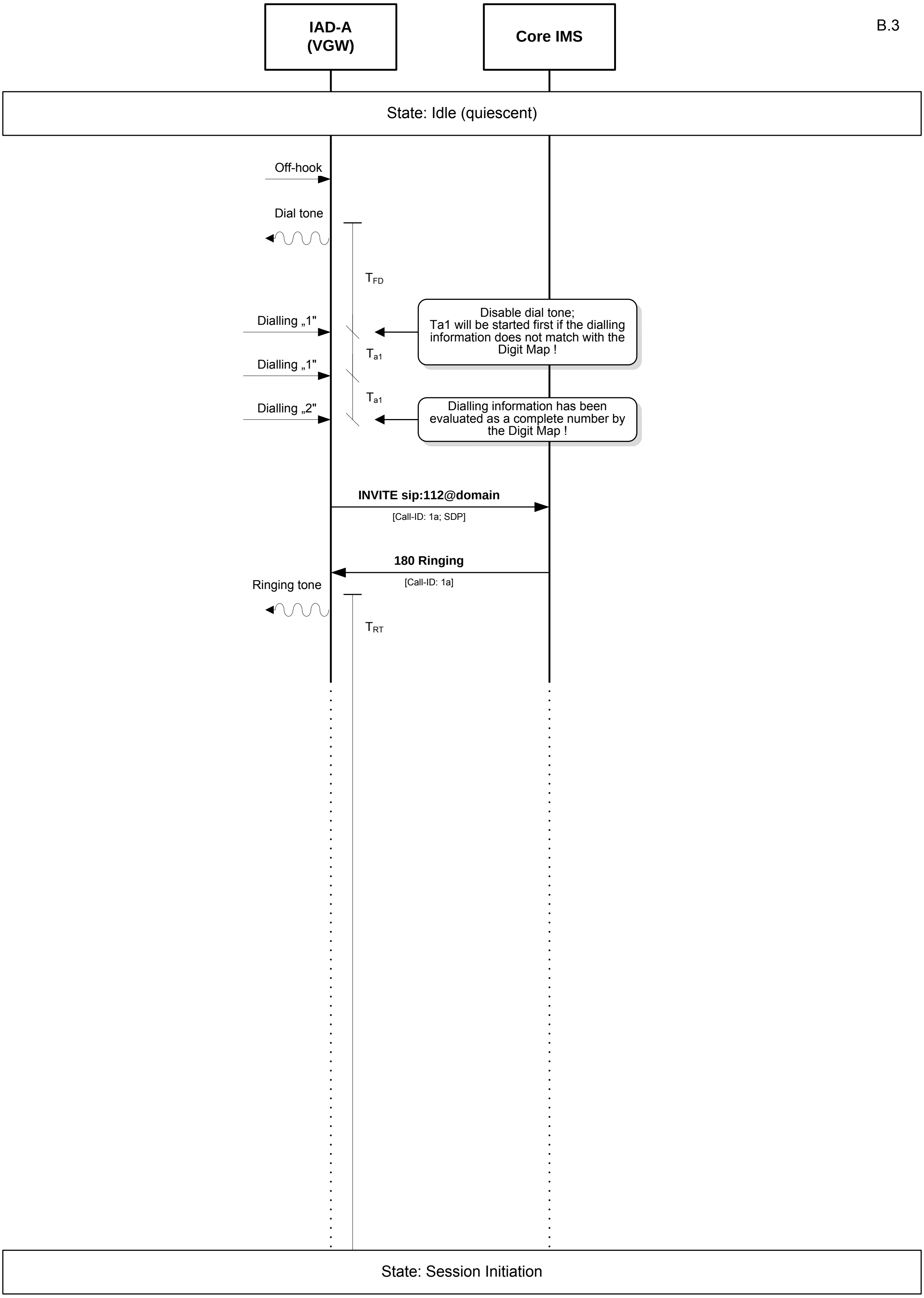
B.1 Normal session attempt



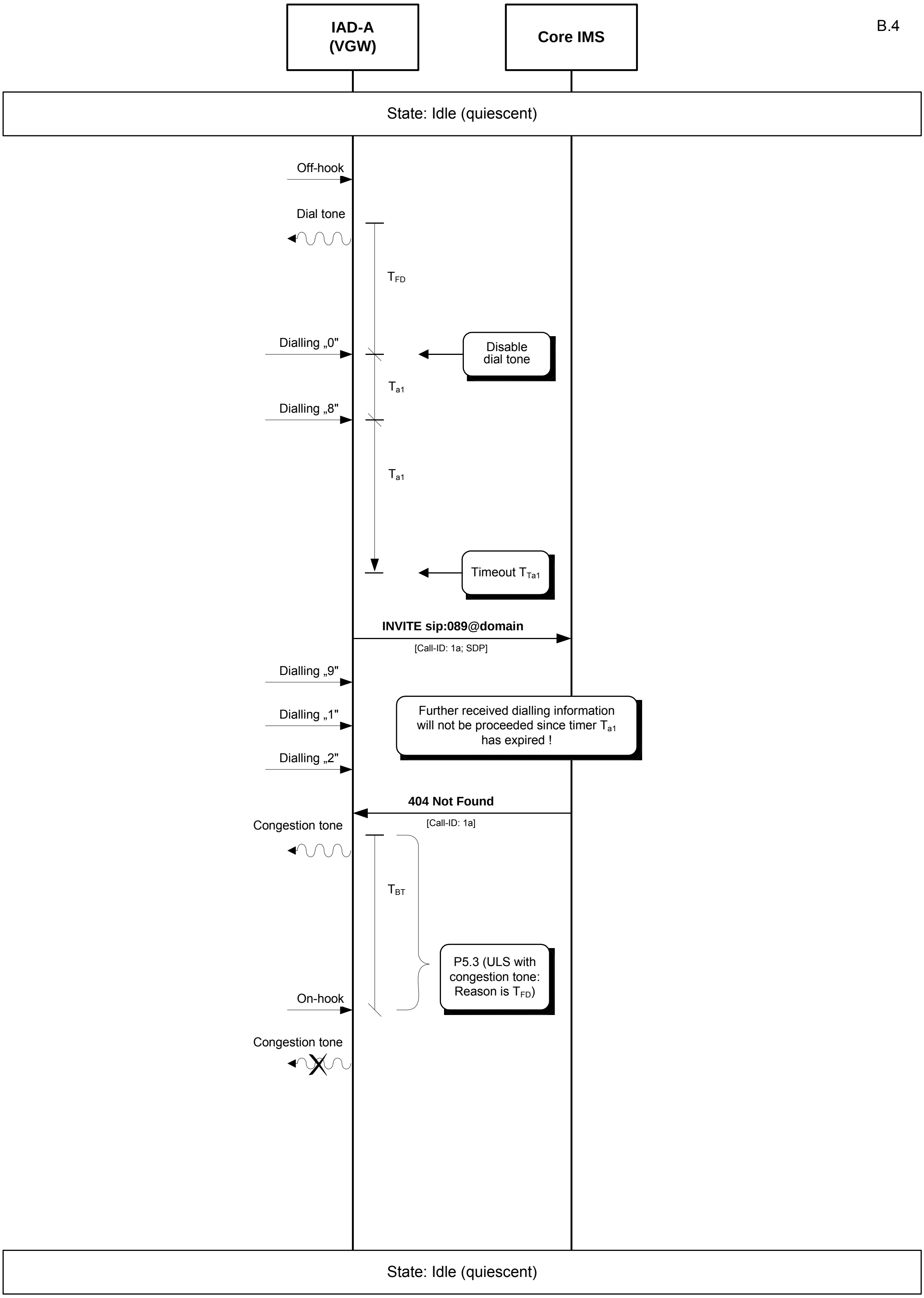
B.2 Off-hook without dialling



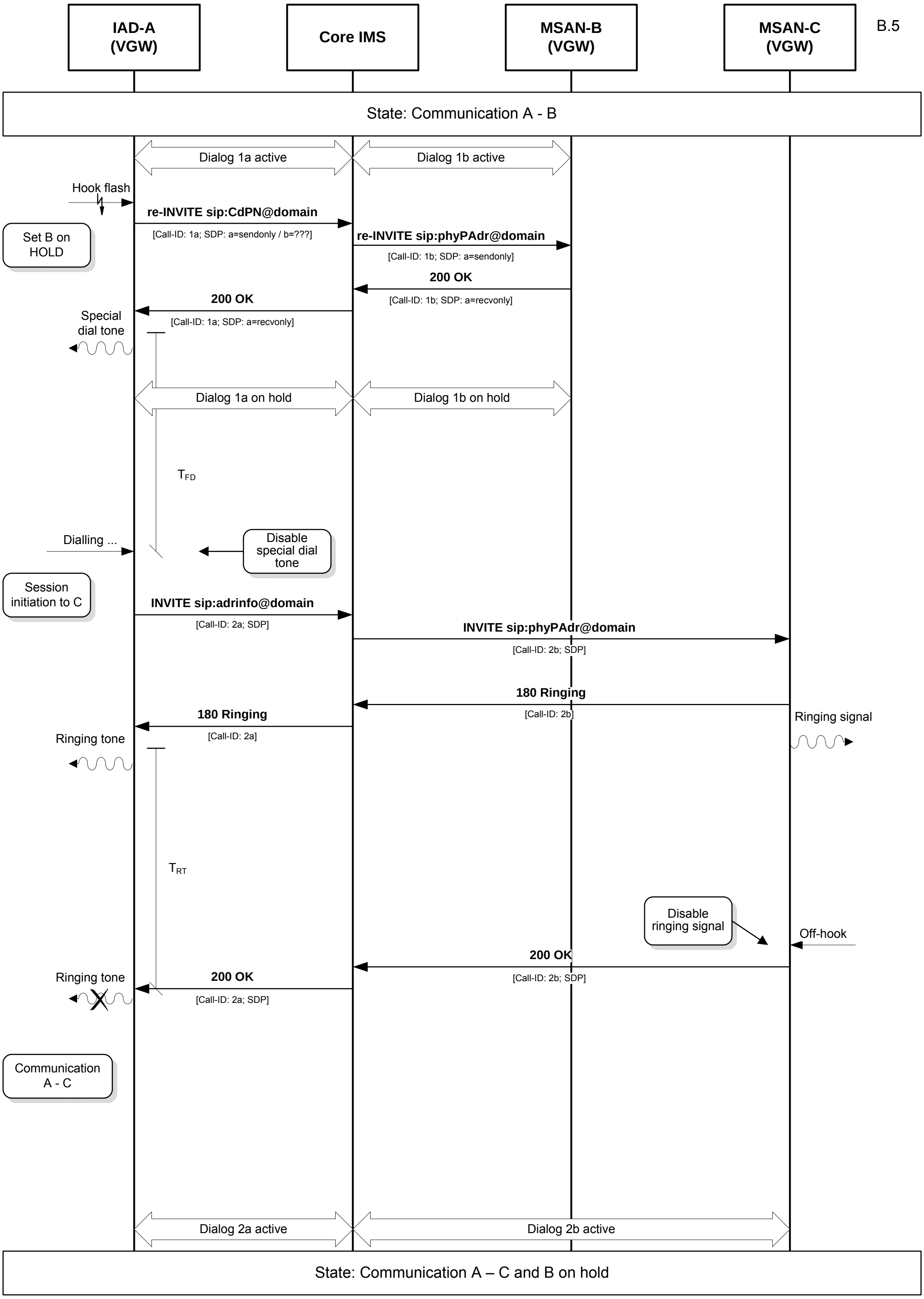
B.3 Dialling, with use of Digit-Map (Emergency Communication)



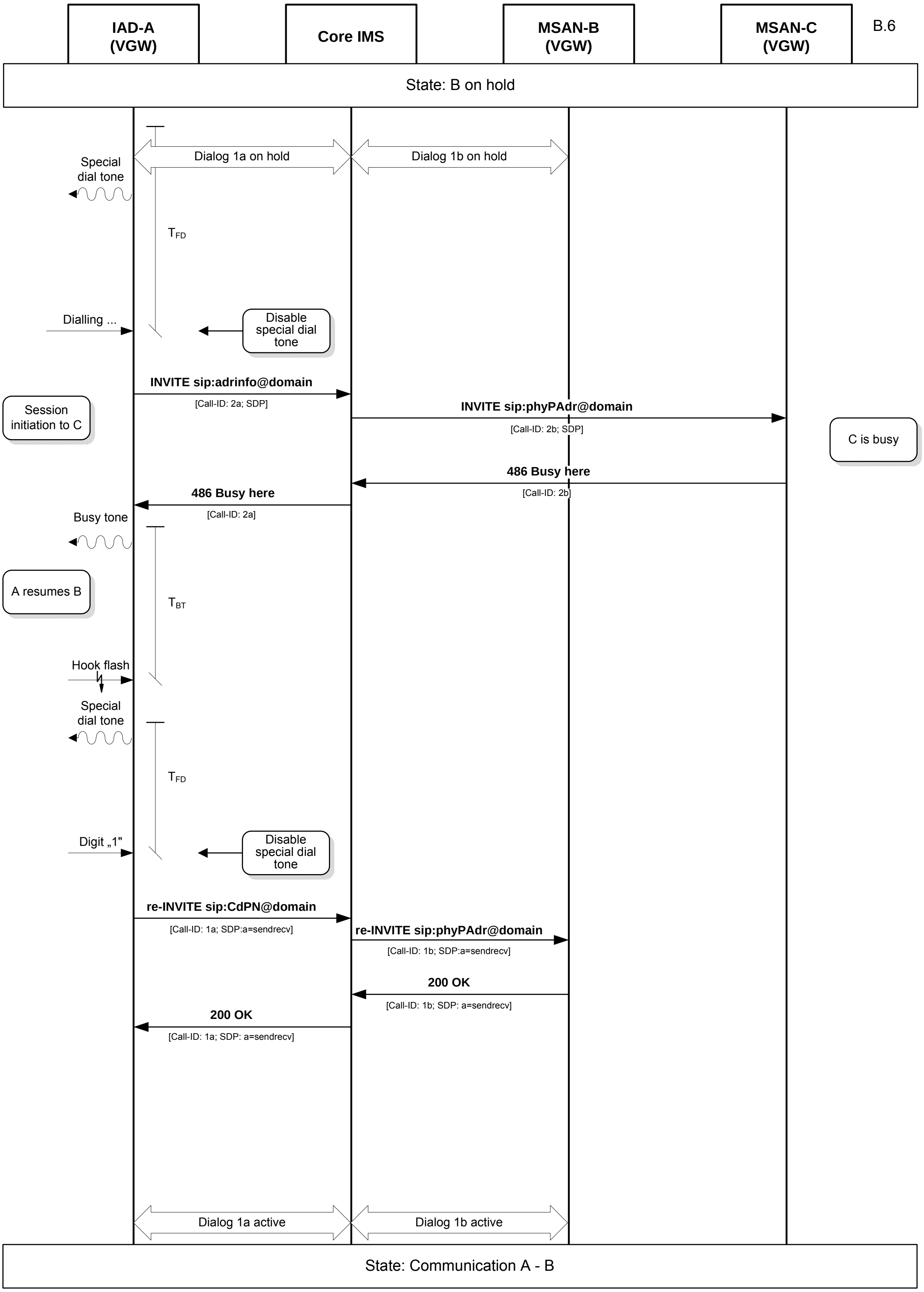
B.4 Dialling with expiry of Inter-Digital Pause timer ($T_{IDP} = T_{a1}$)



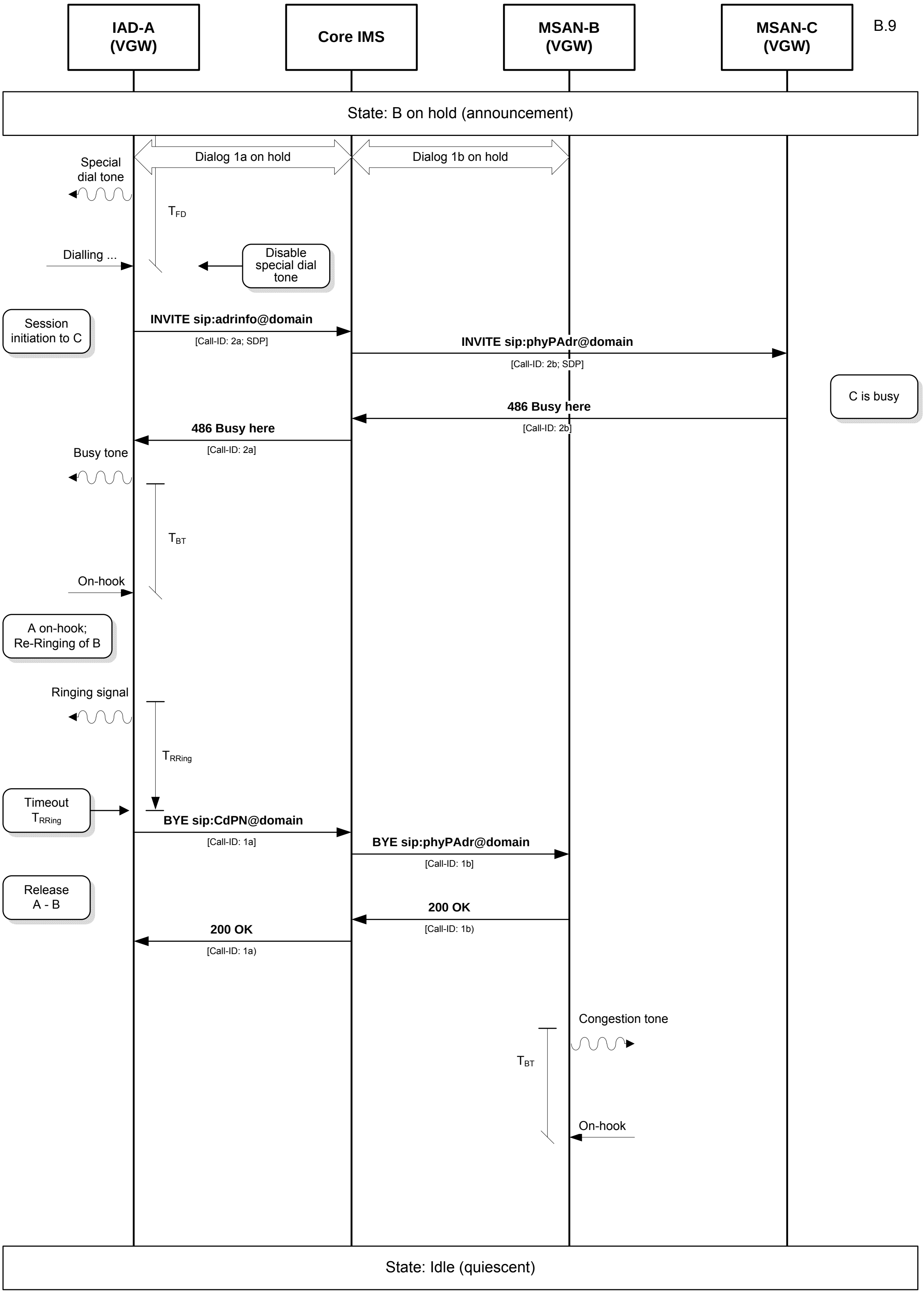
B.5 Toggle: Invoke HOLD and Toggle



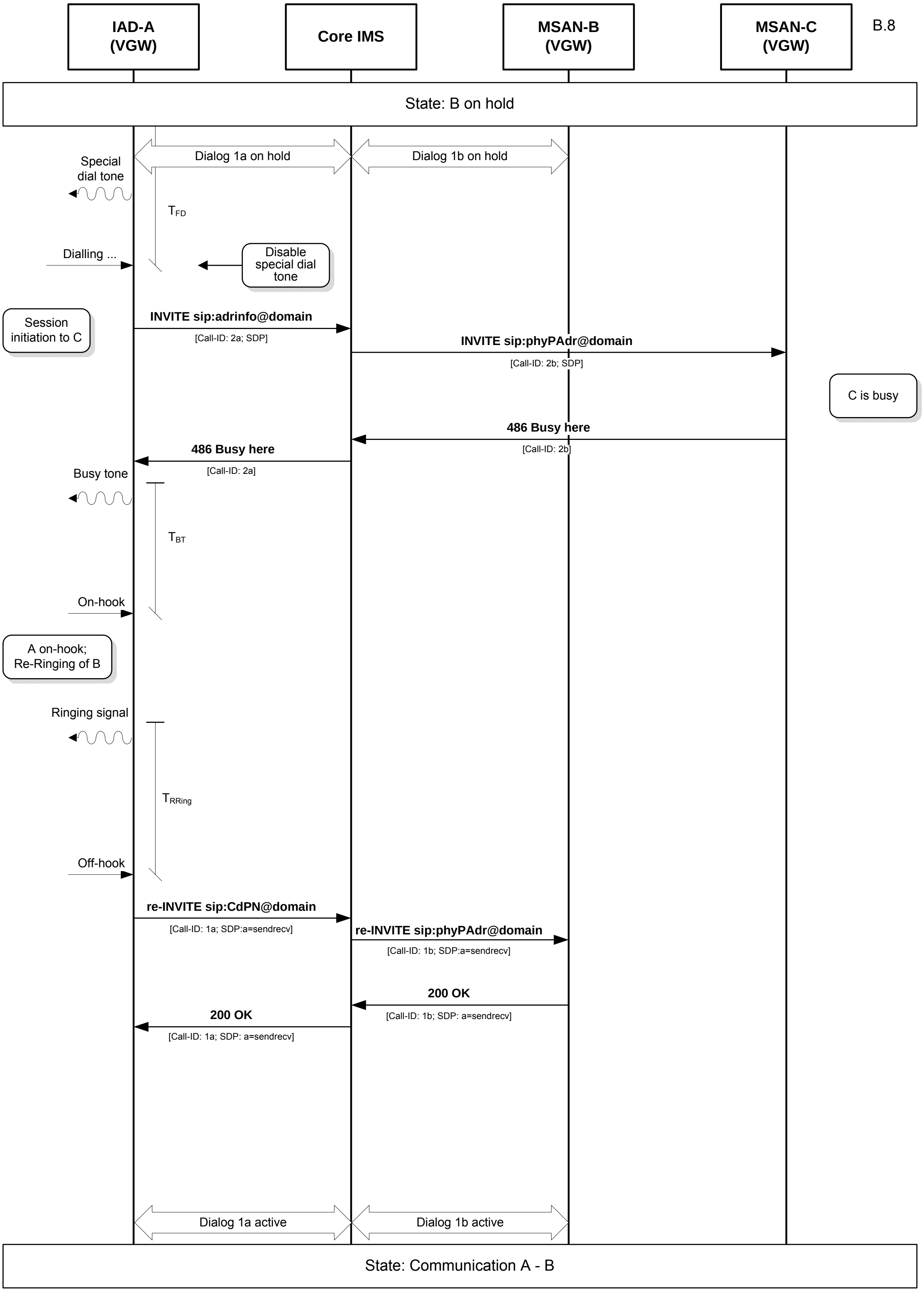
B.6 Toggle: Destination busy, resume with hook-flash



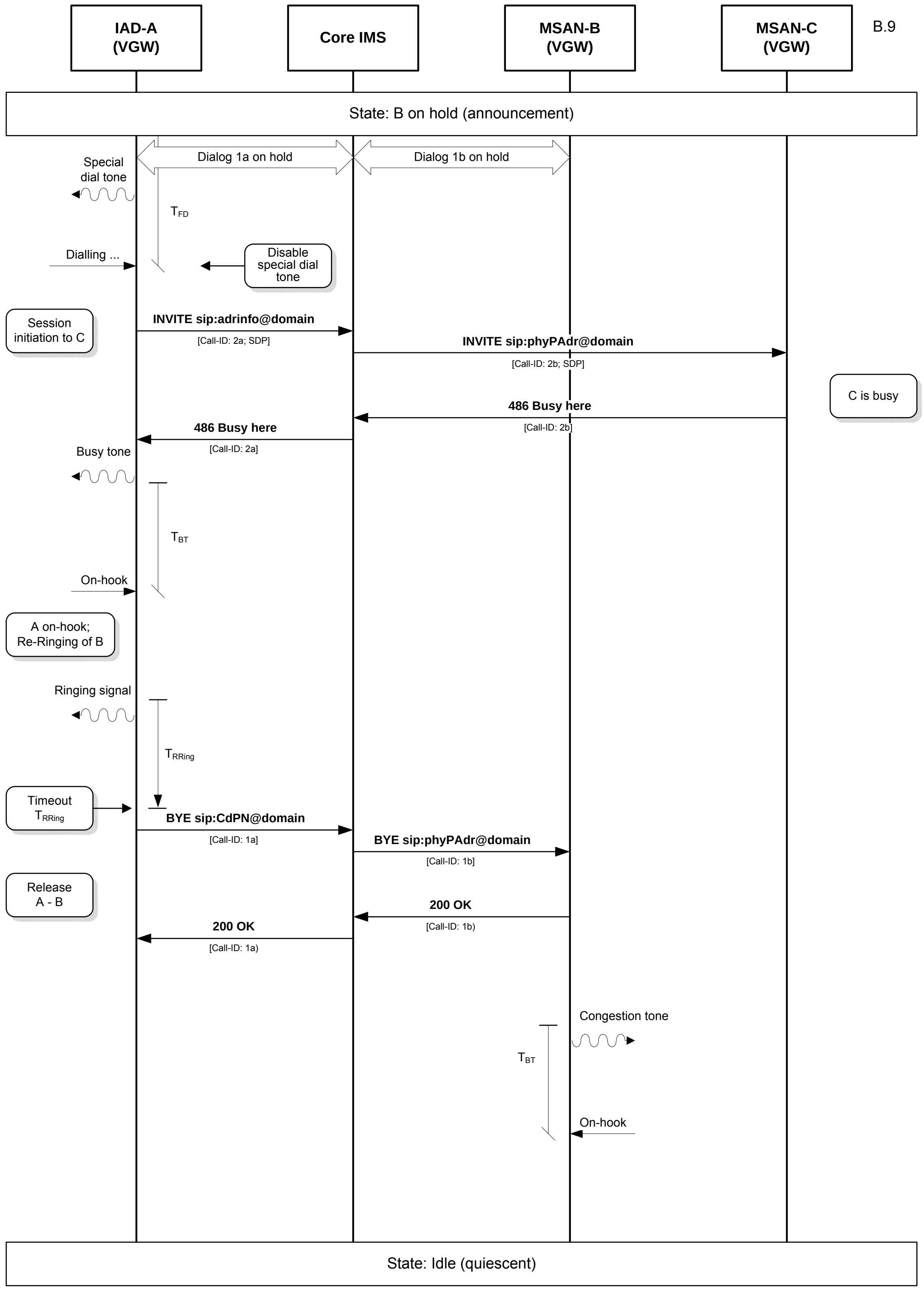
B.7 Toggle: Abort toggle, resume with hook-flash



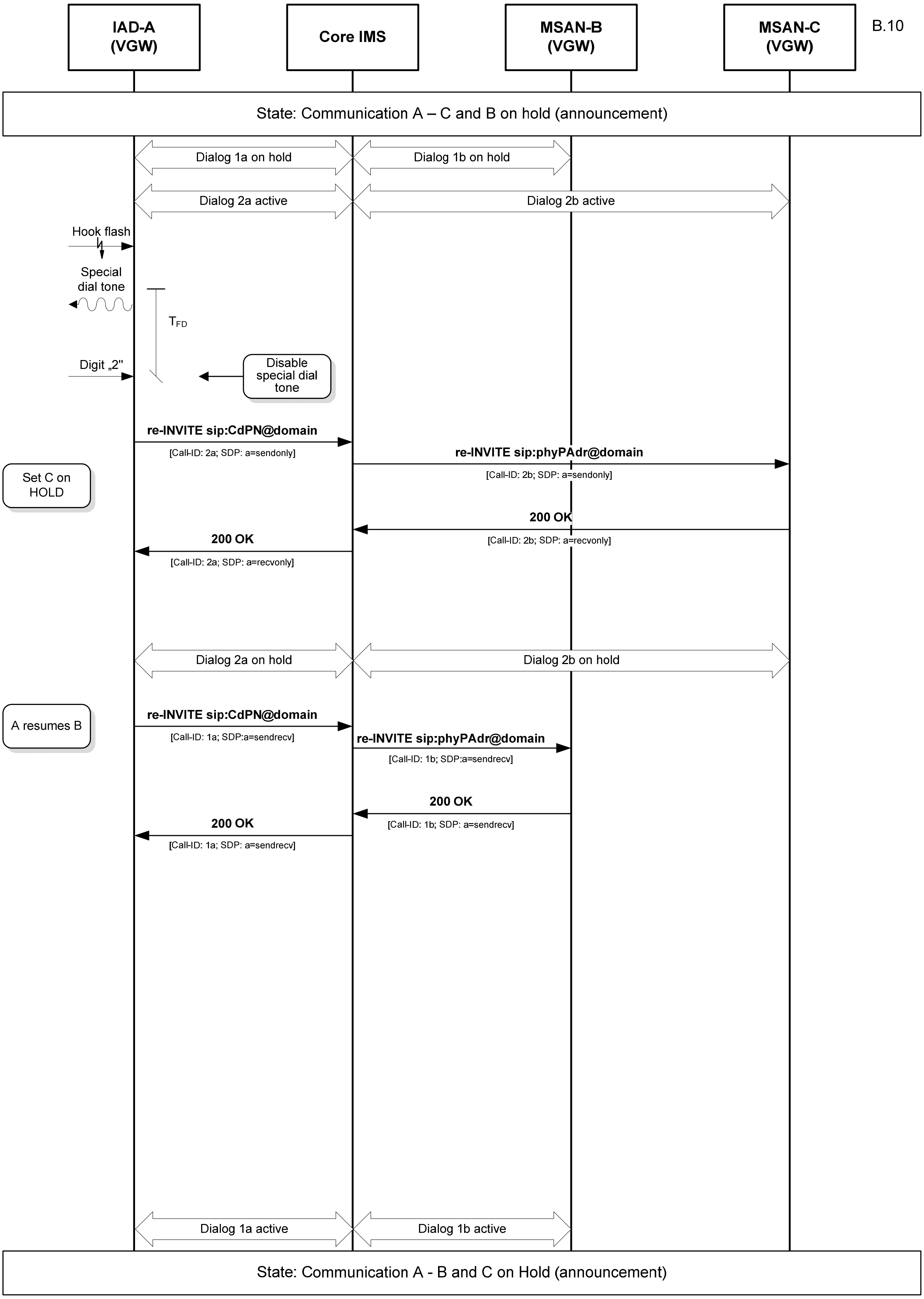
B.8 Toggle: Destination busy, resume with Re-Ringing



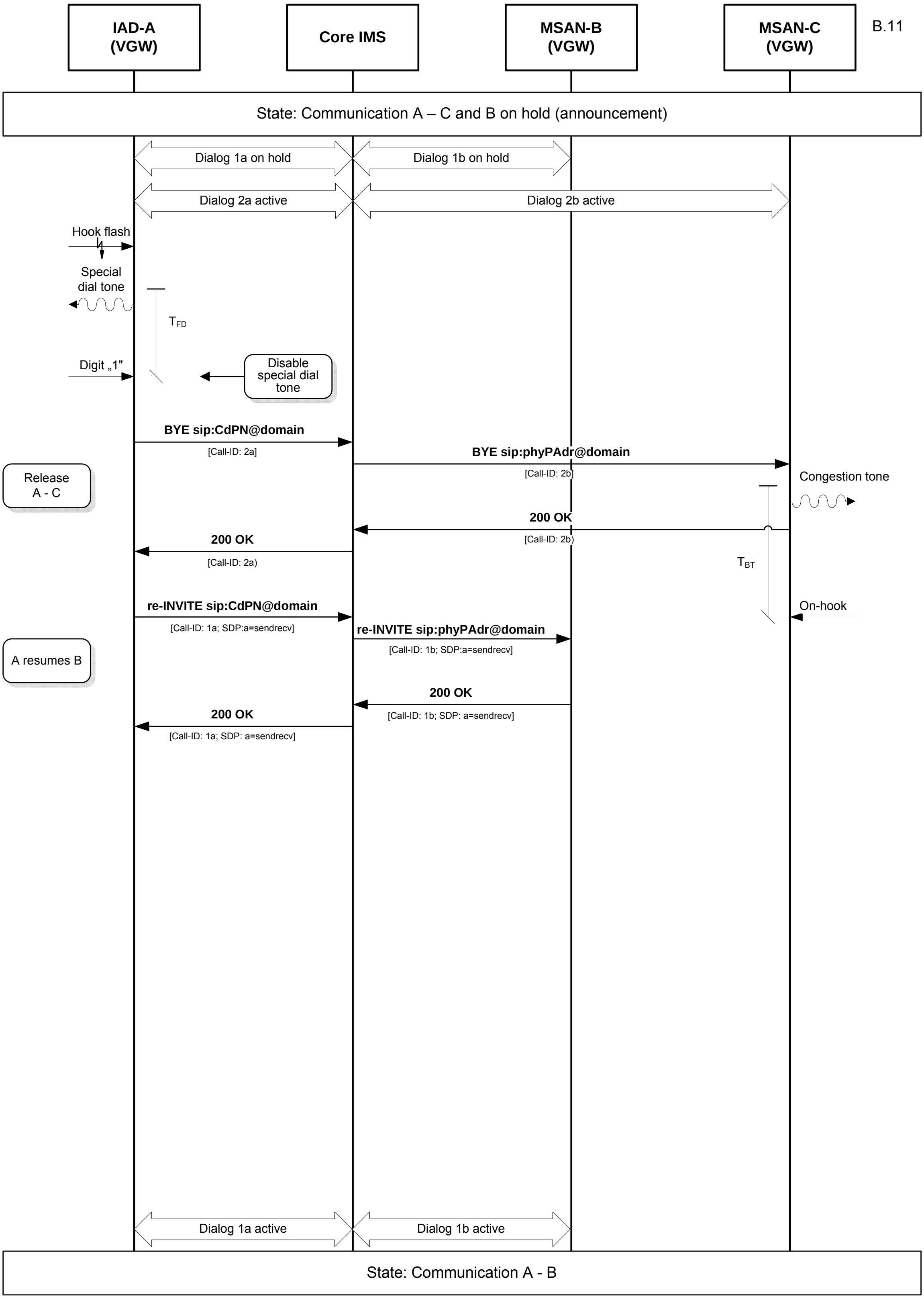
B.9 Toggle: Destination busy, Re-Ringing ignored



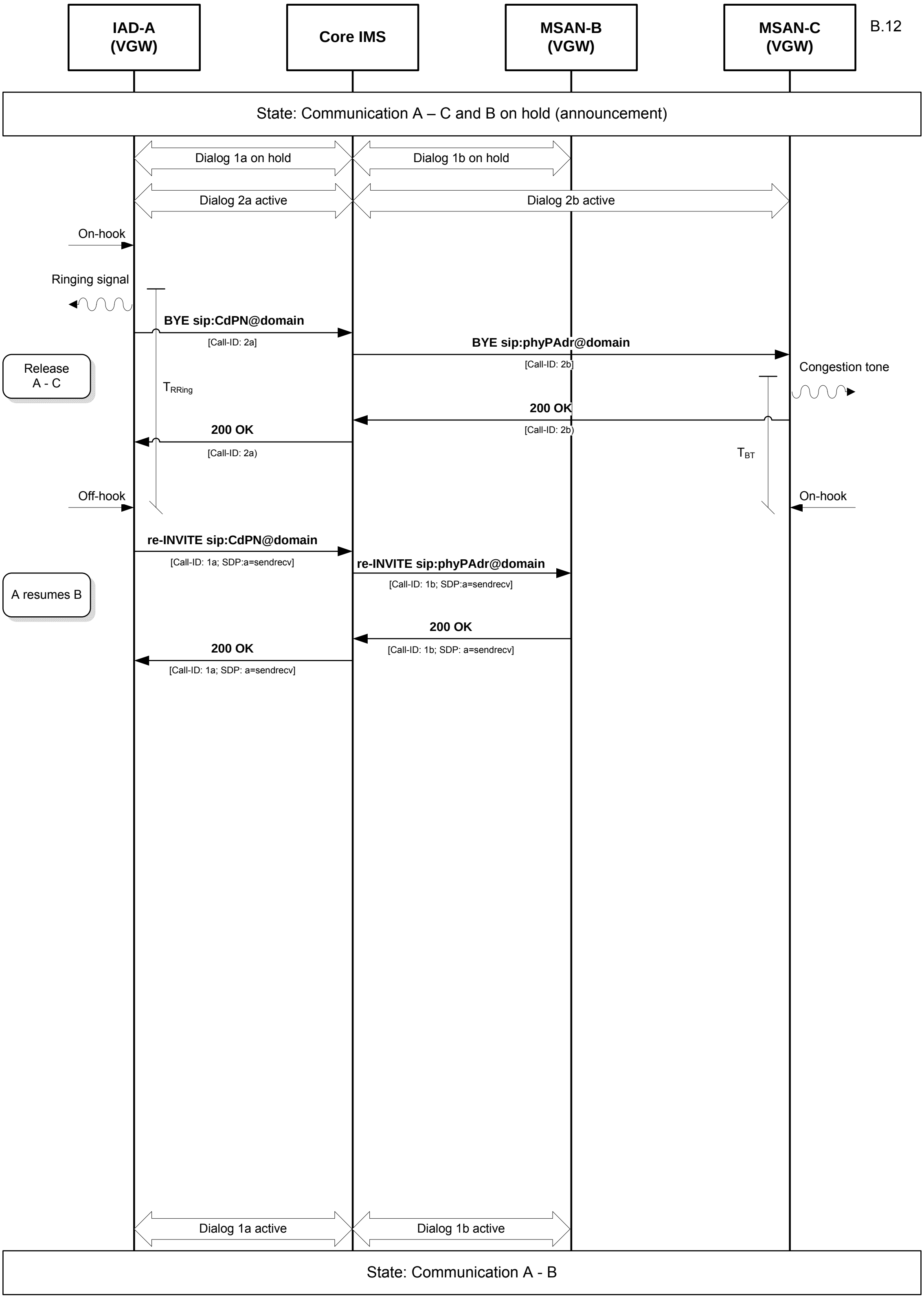
B.10 Toggle: Set C on HOLD and toggle with B



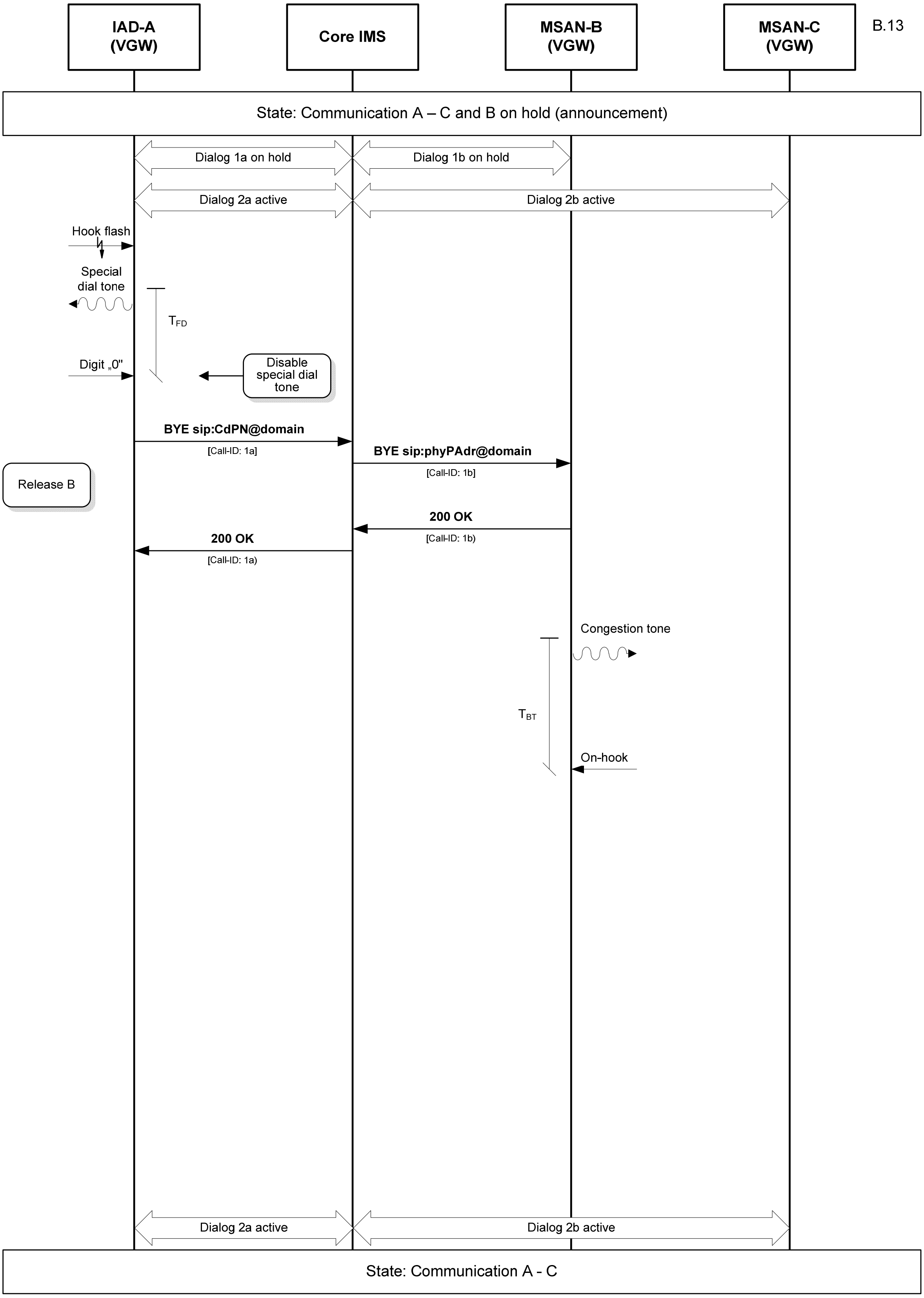
B.11 Toggle: Release C and toggle to B using SOC



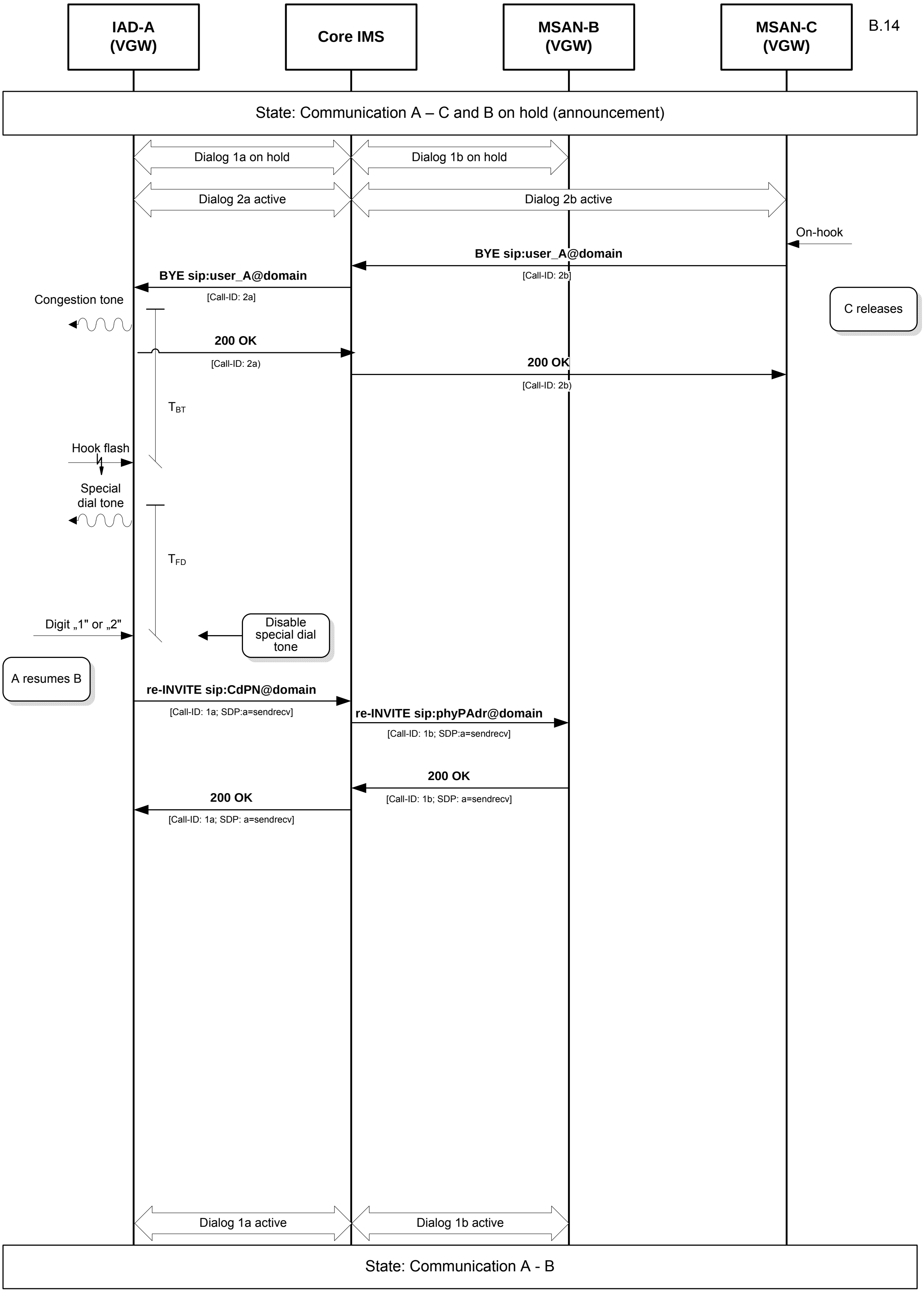
B.12 Toggle: Release C and resume B with Re-Ringing procedure



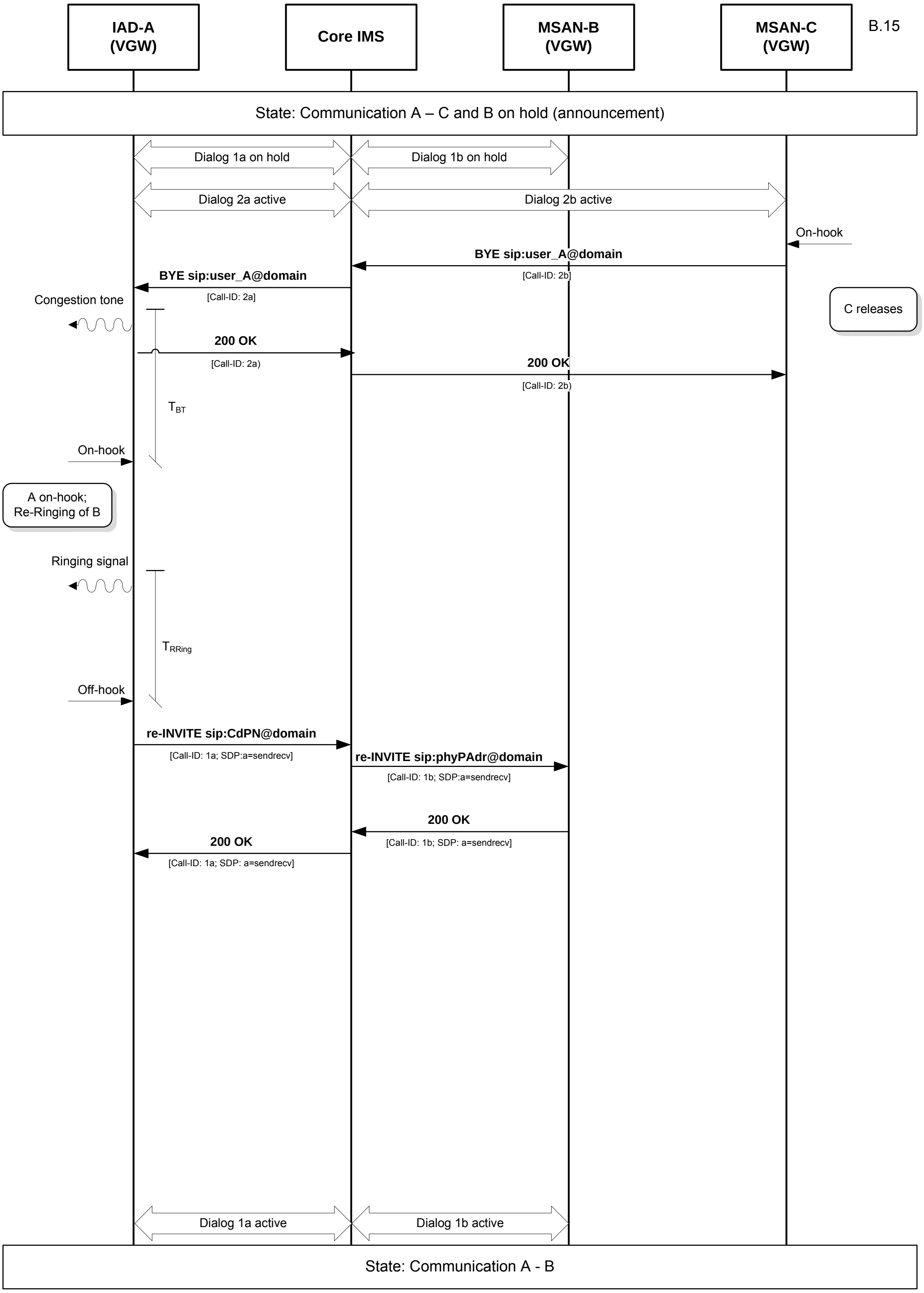
B.13 Toggle: Release B using SOC



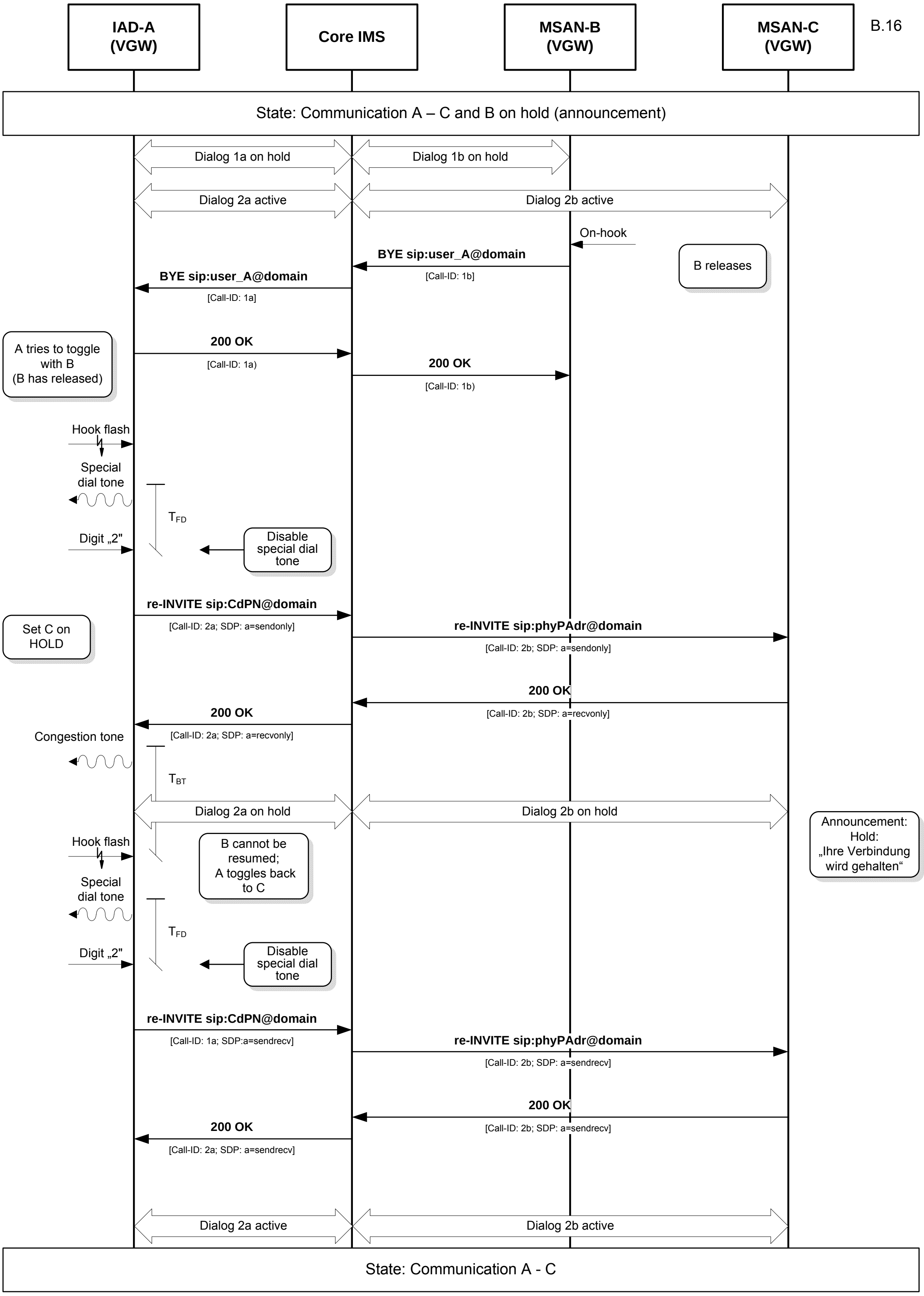
B.14 Toggle: C releases communication and resume B using SOC



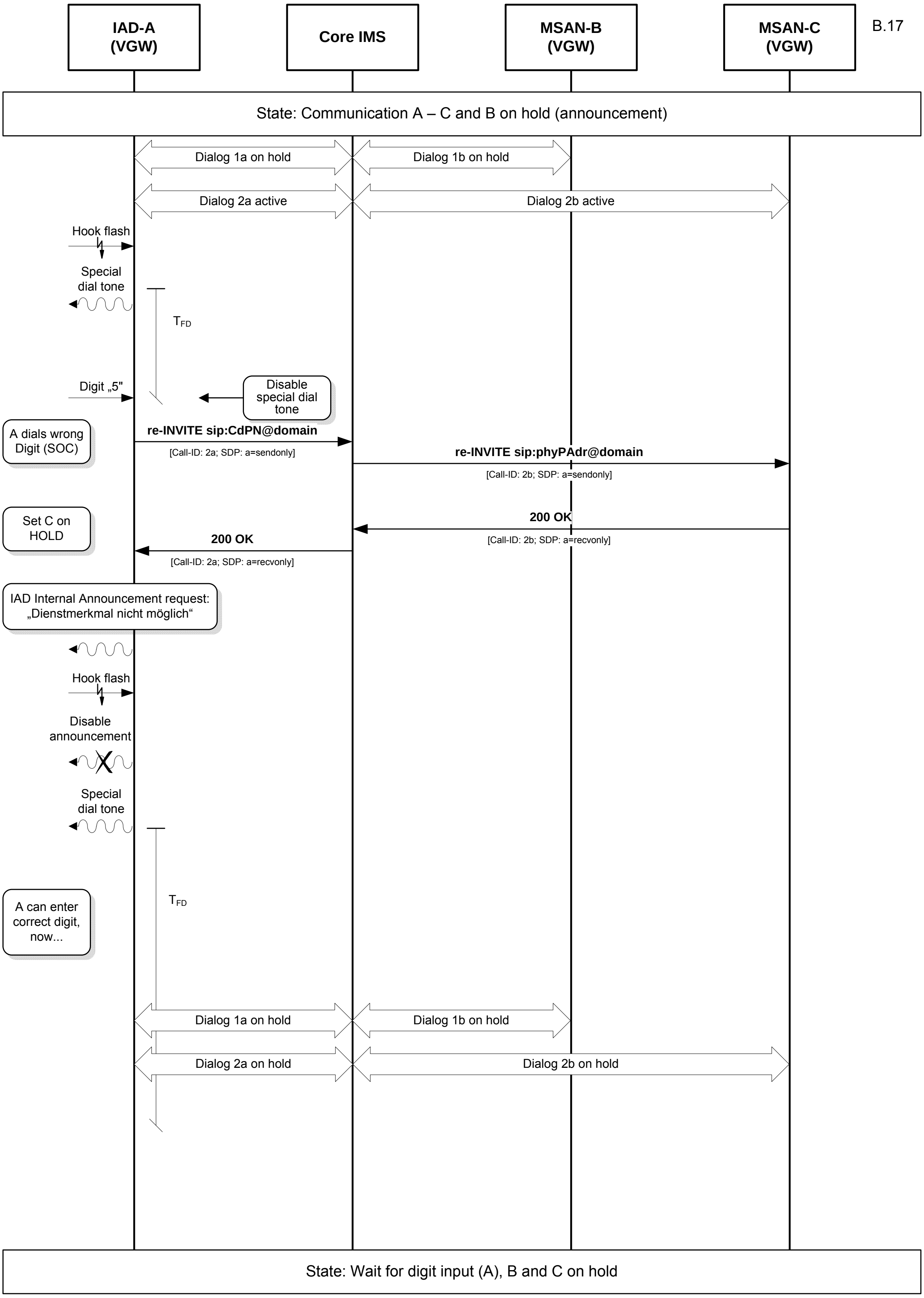
B.15 Toggle: C releases communication and resume B using Re-Ringing procedure



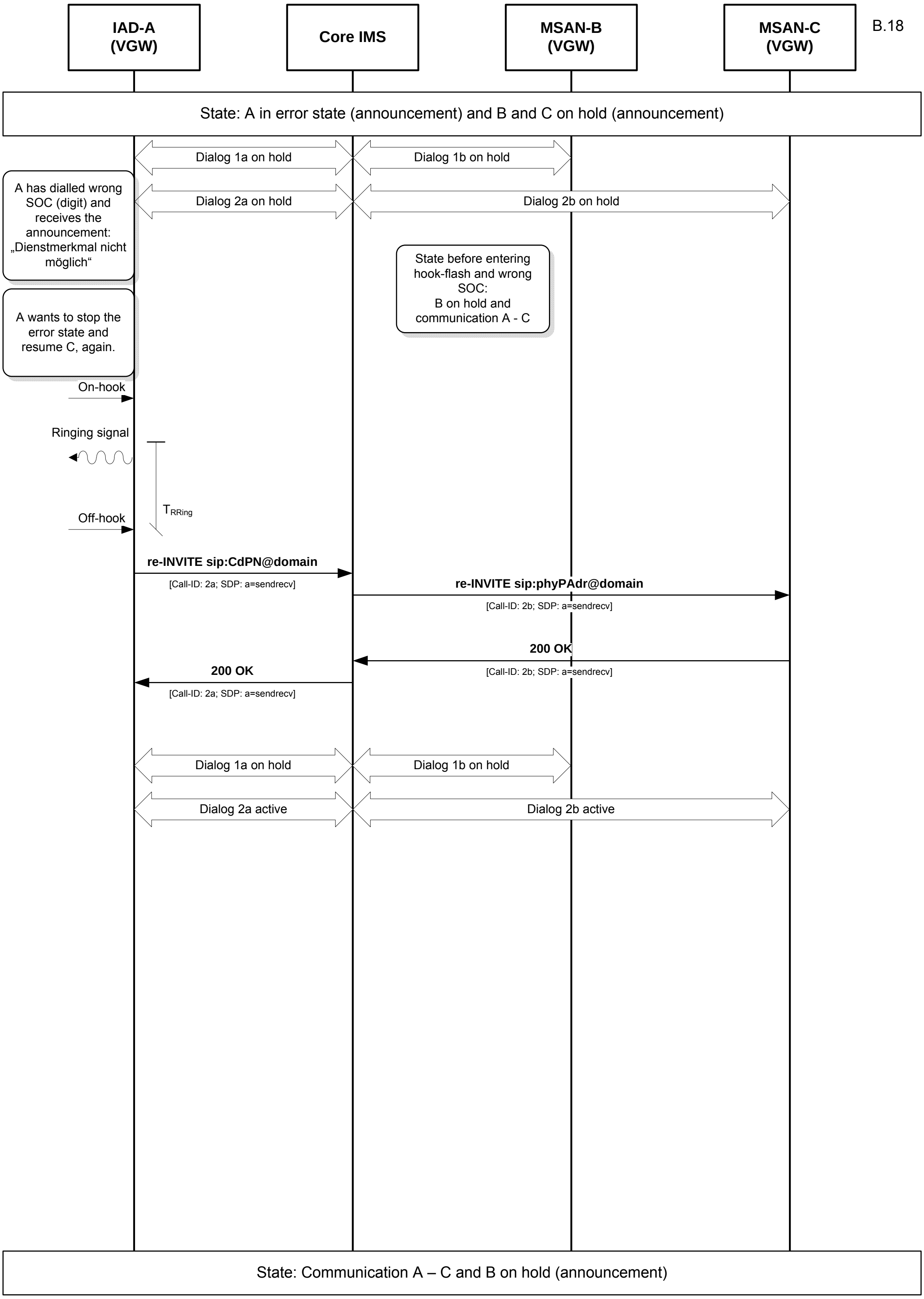
B.16 Toggle: B on HOLD releases, resume procedure failed



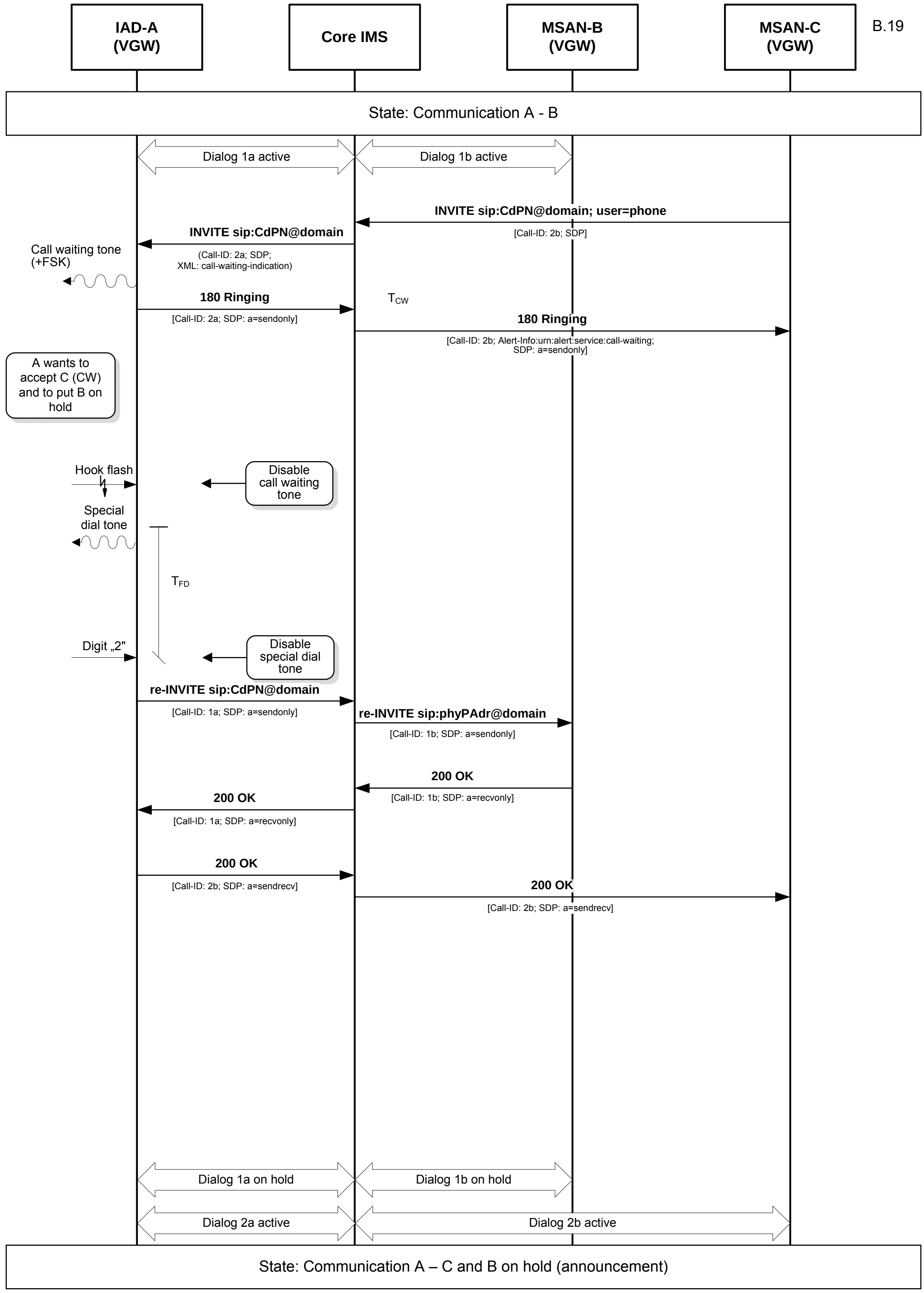
B.17 Toggle: Toggle, using wrong SOC and hook-flash



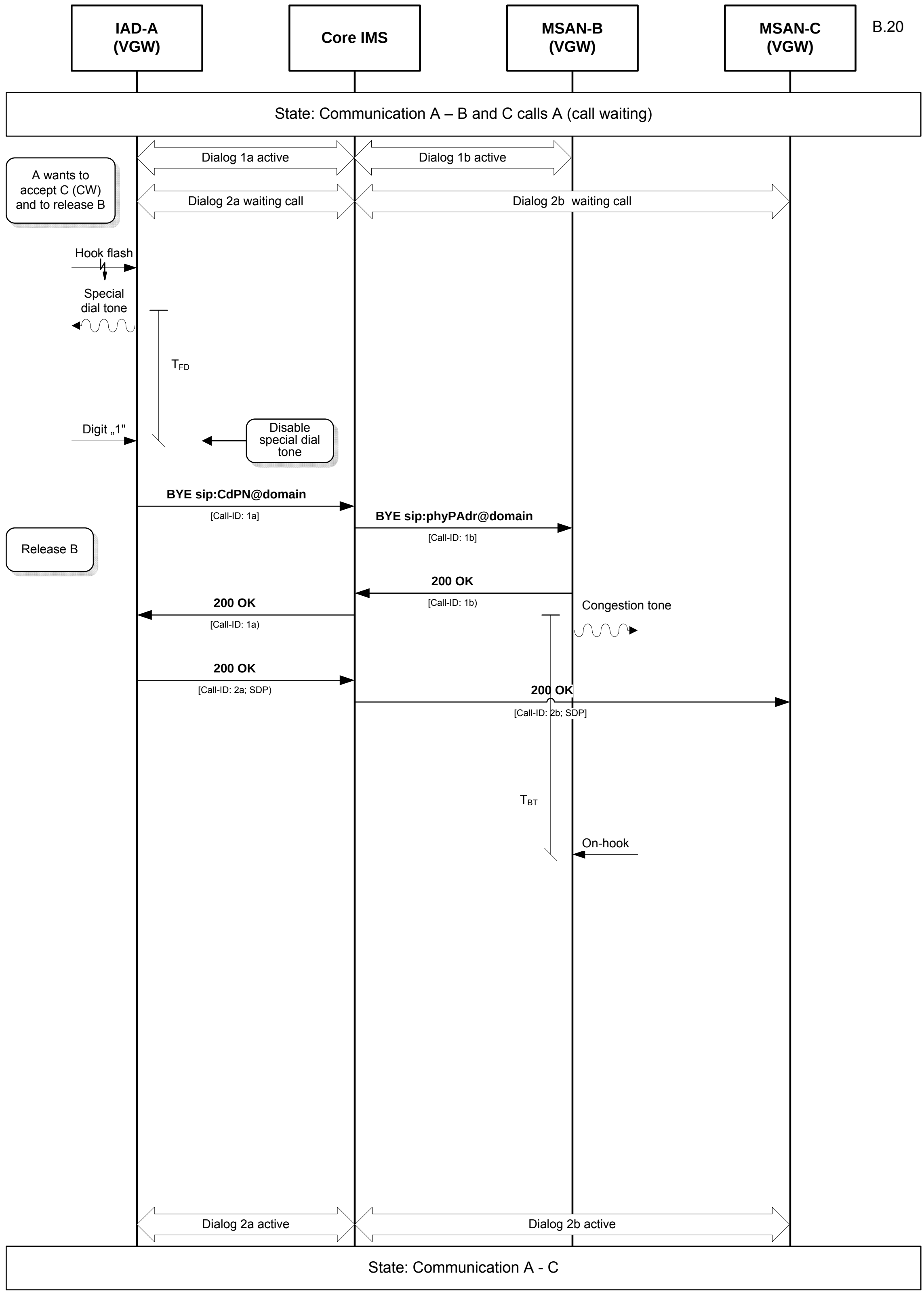
B.18 Toggle: Toggle, using wrong SOC and Re-Ringing



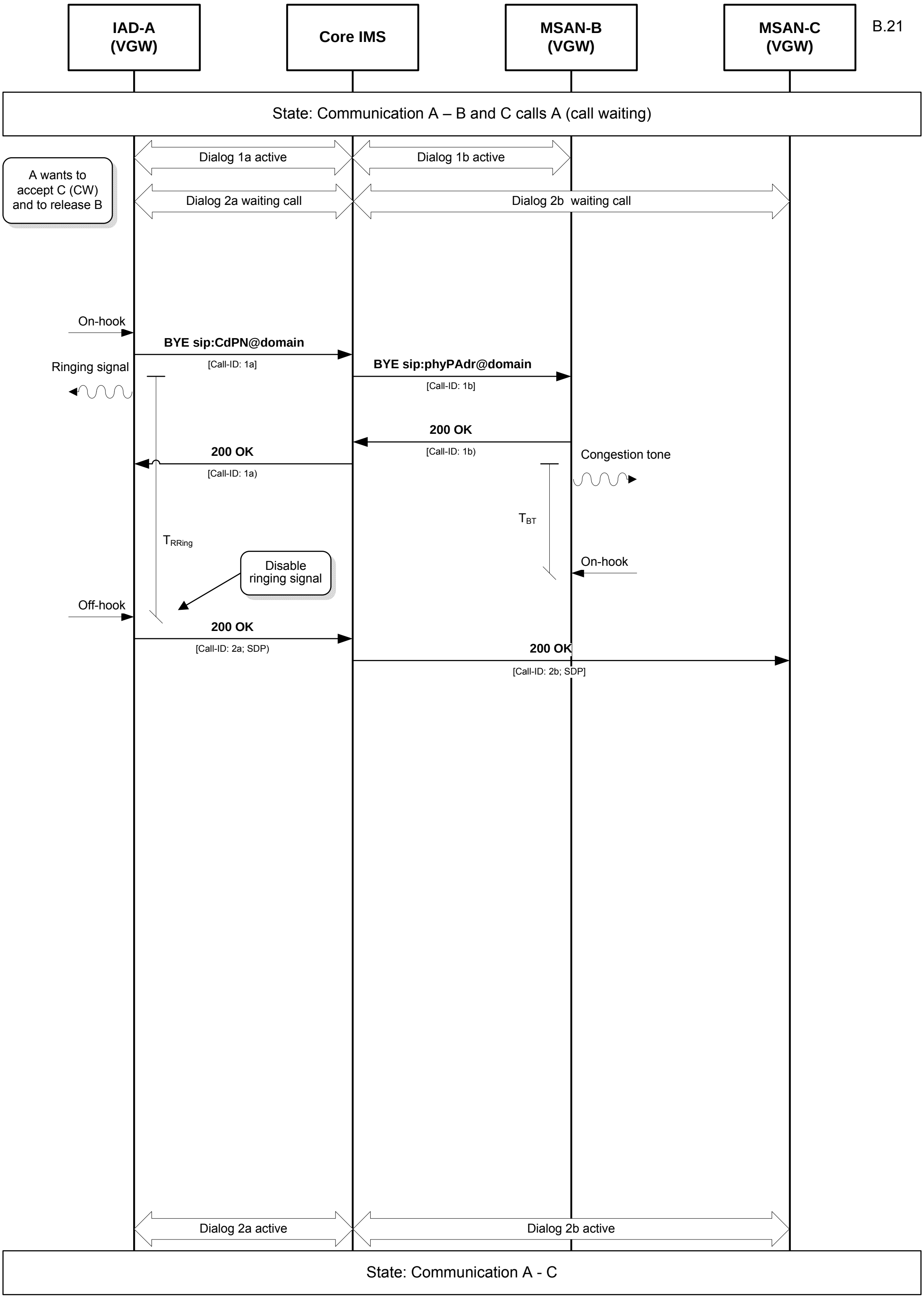
B.19 CW: Accept waiting communication of C and set B on HOLD using SOC



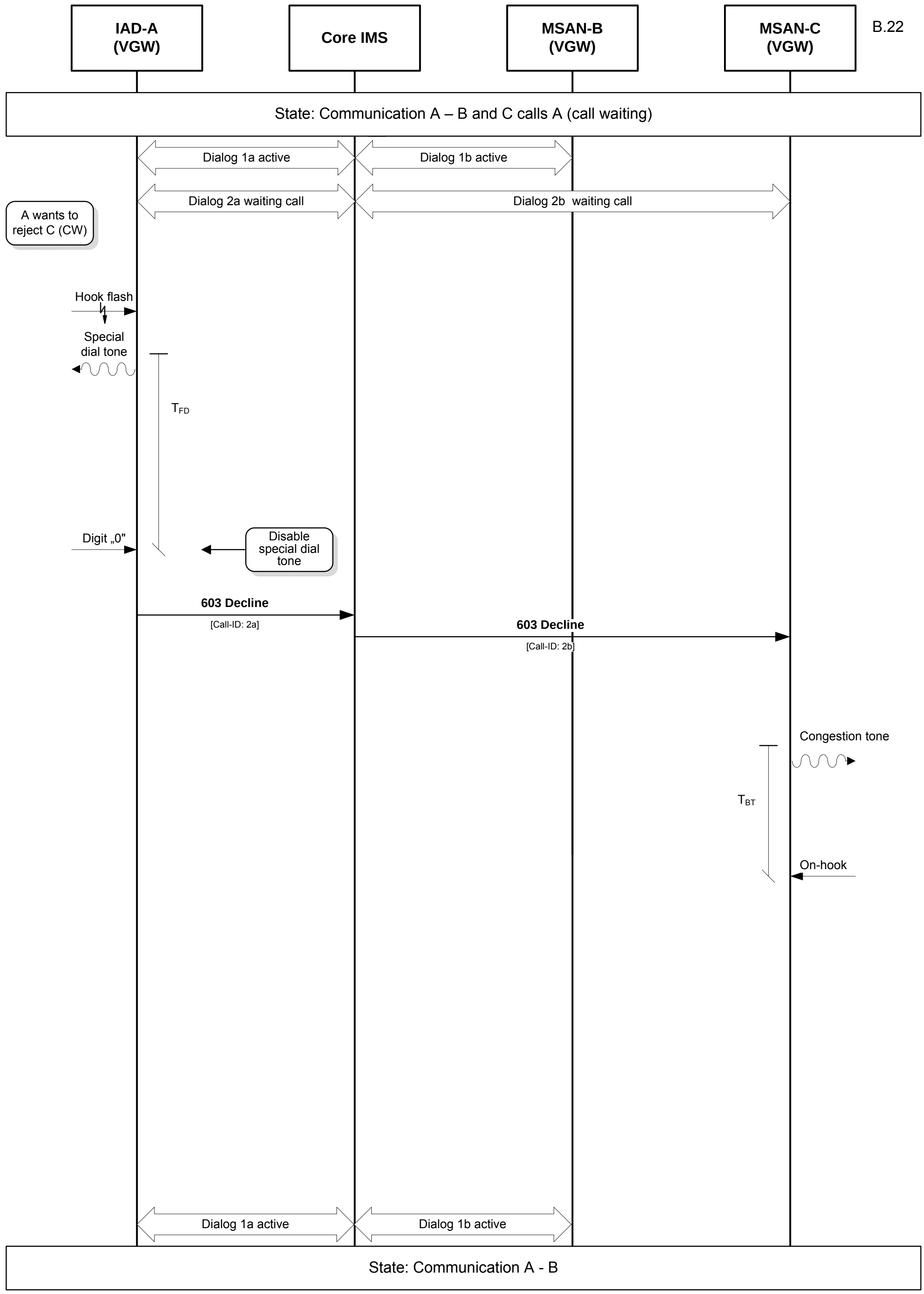
B.20 CW: Accept waiting communication and release B using SOC



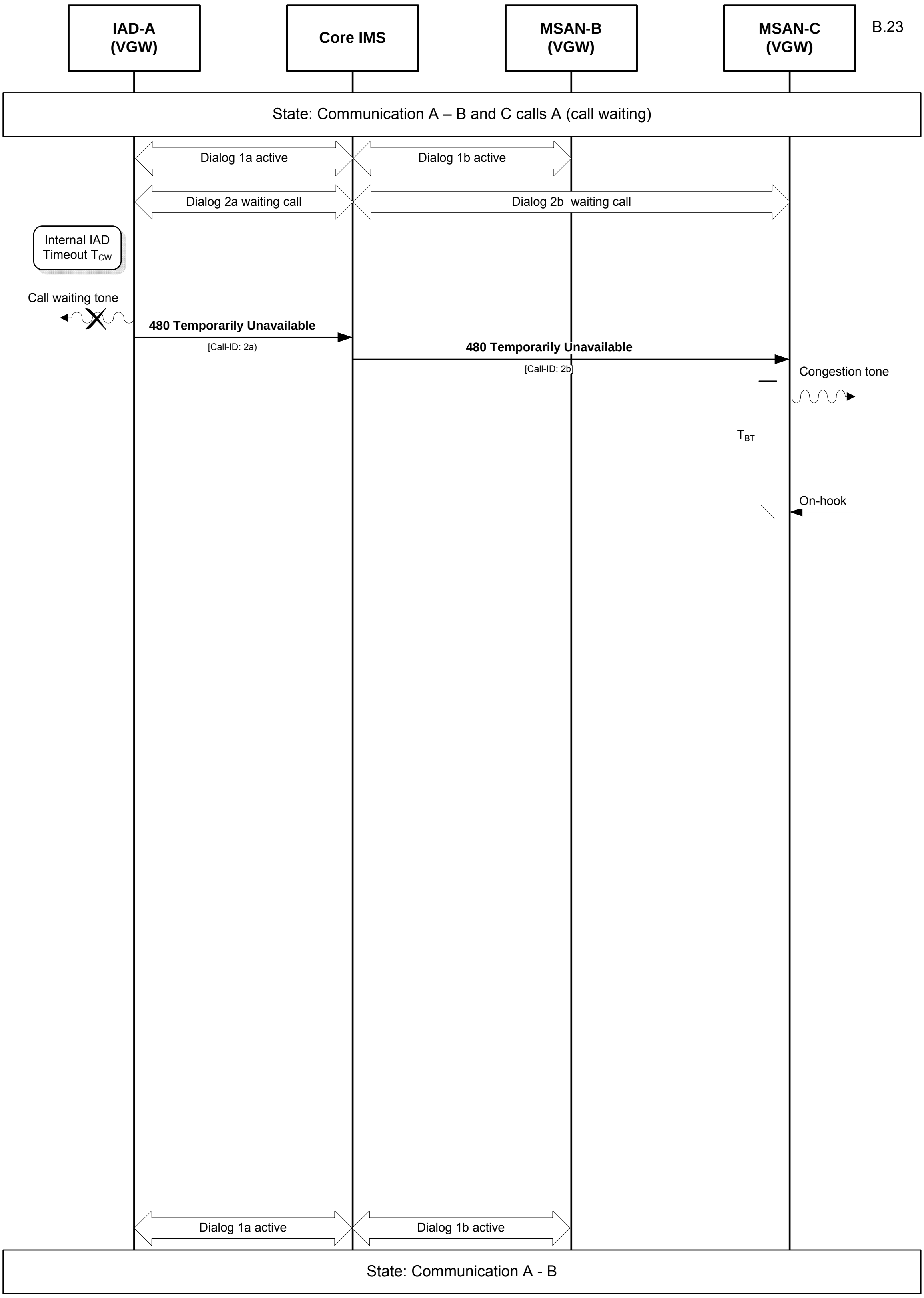
B.21 CW: Accept waiting communication of C using Re-Ringing



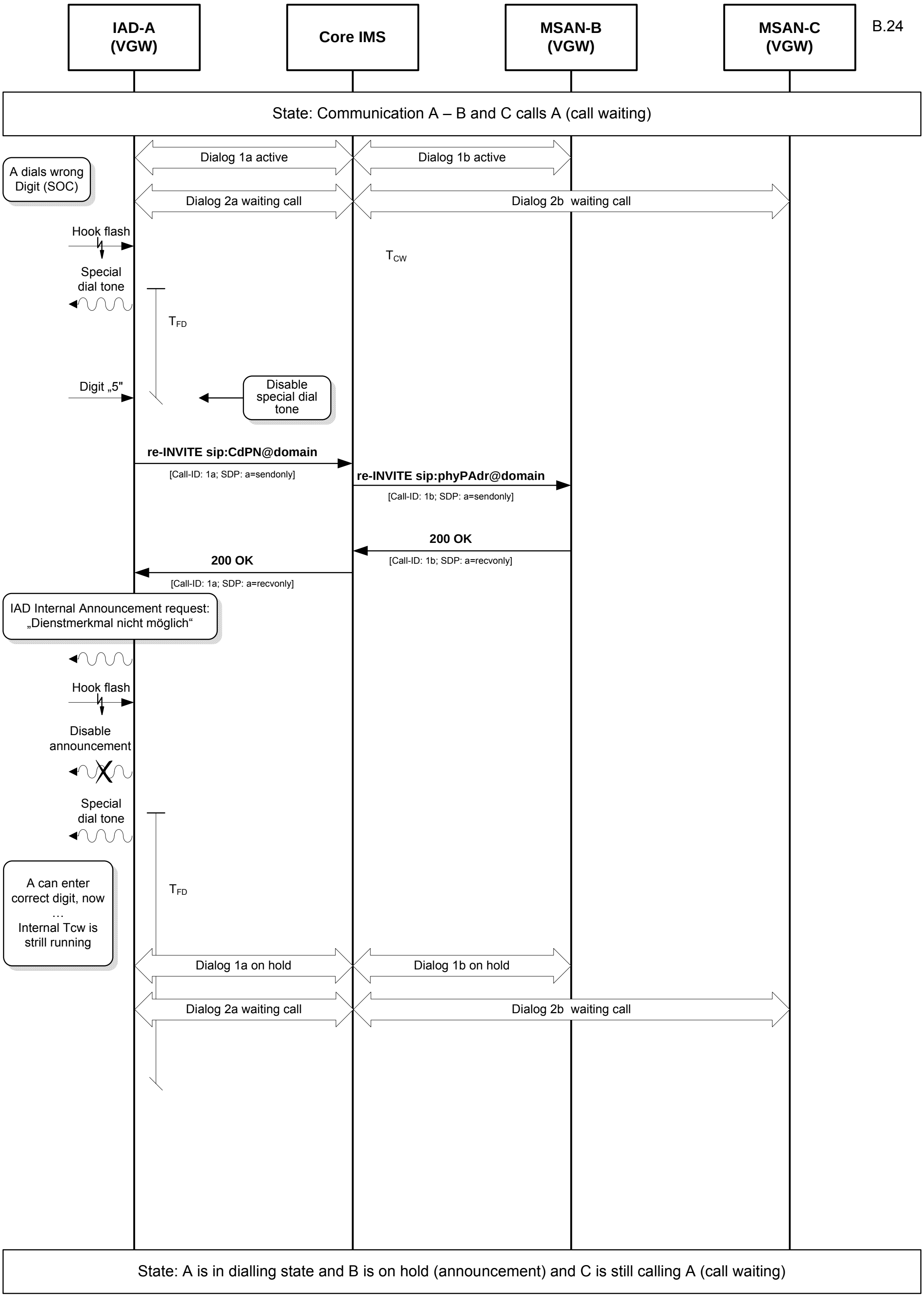
B.22 CW: Reject waiting communication



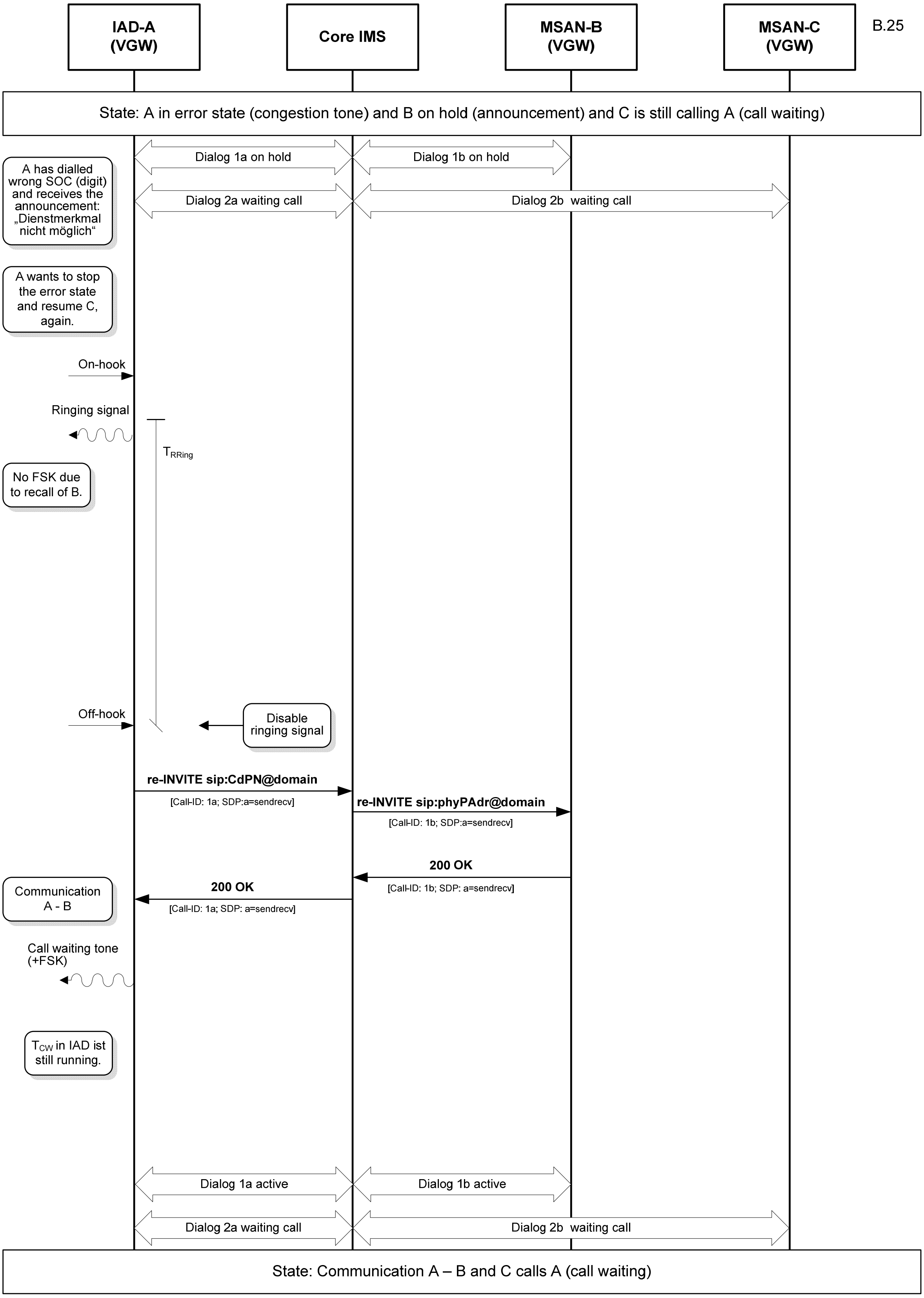
B.23 CW: Ignore waiting communication



B.24 CW: Communication waiting, wrong SOC and Hook-flash

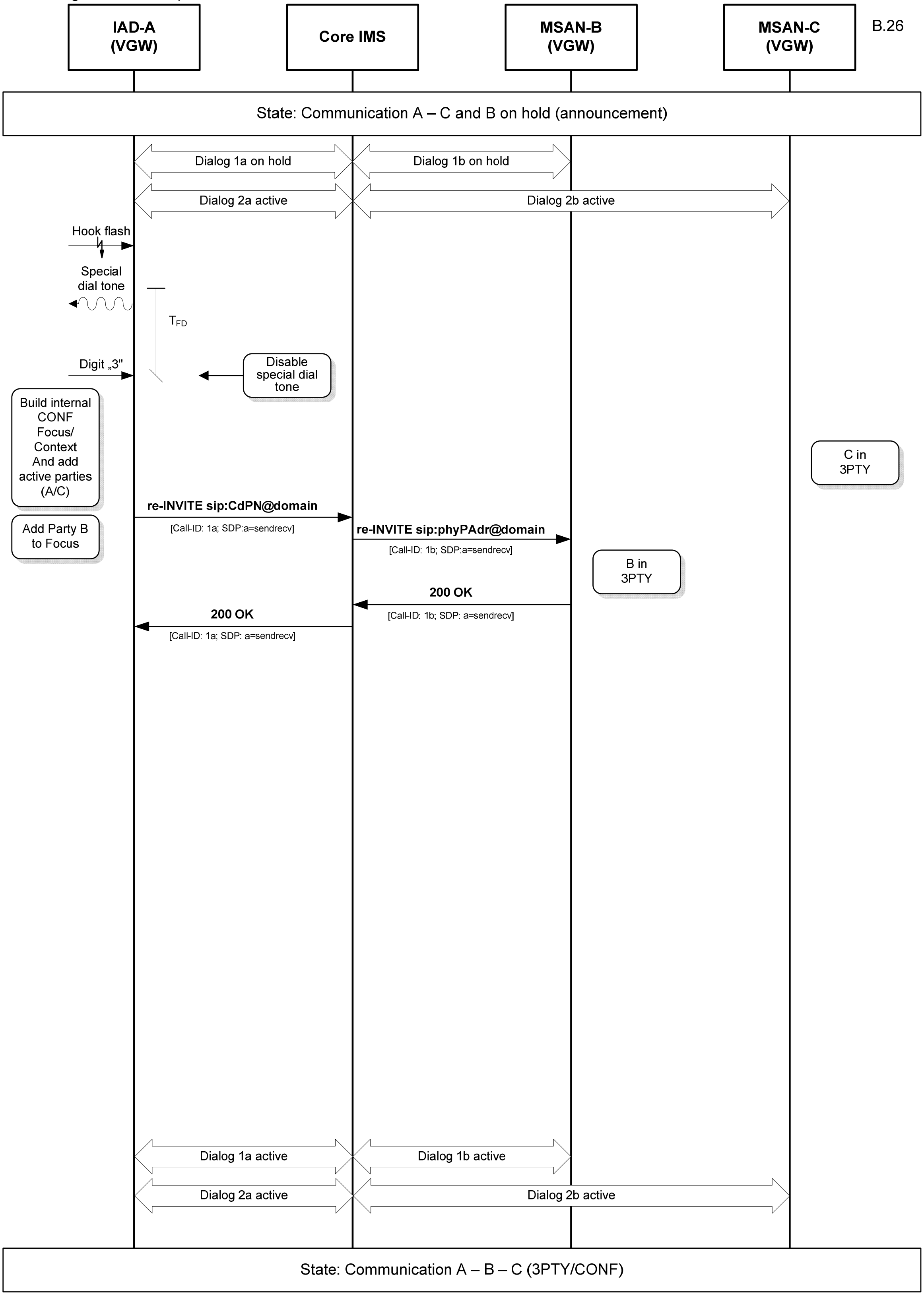


B.25 CW: Communication waiting, wrong SOC and Re-Ringing



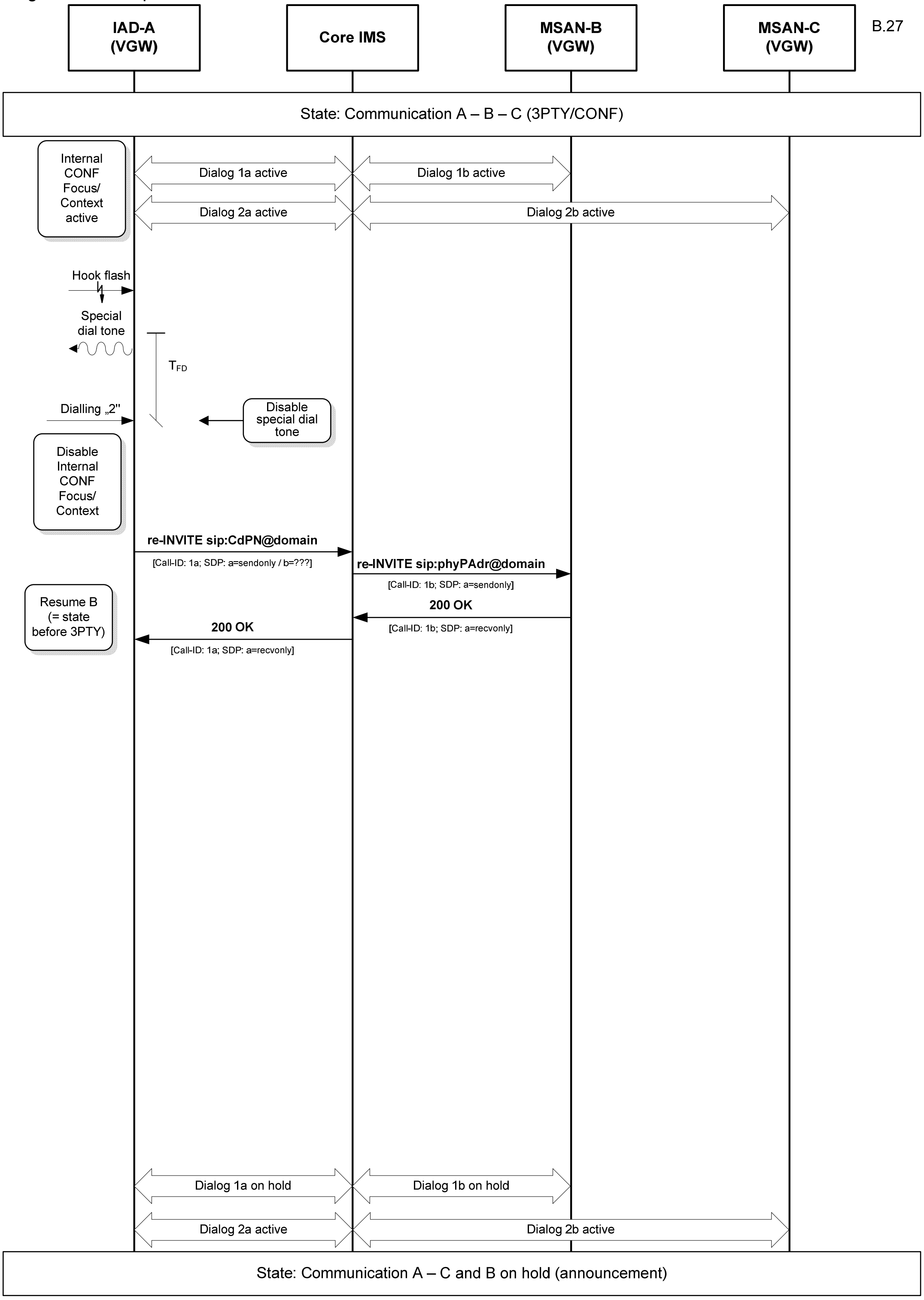
B.26 3PTY: Invoke 3PTY

Conf Bridge must be implemented within the VGW.



B.27 3PTY: Fallback to HOLD/Toggle

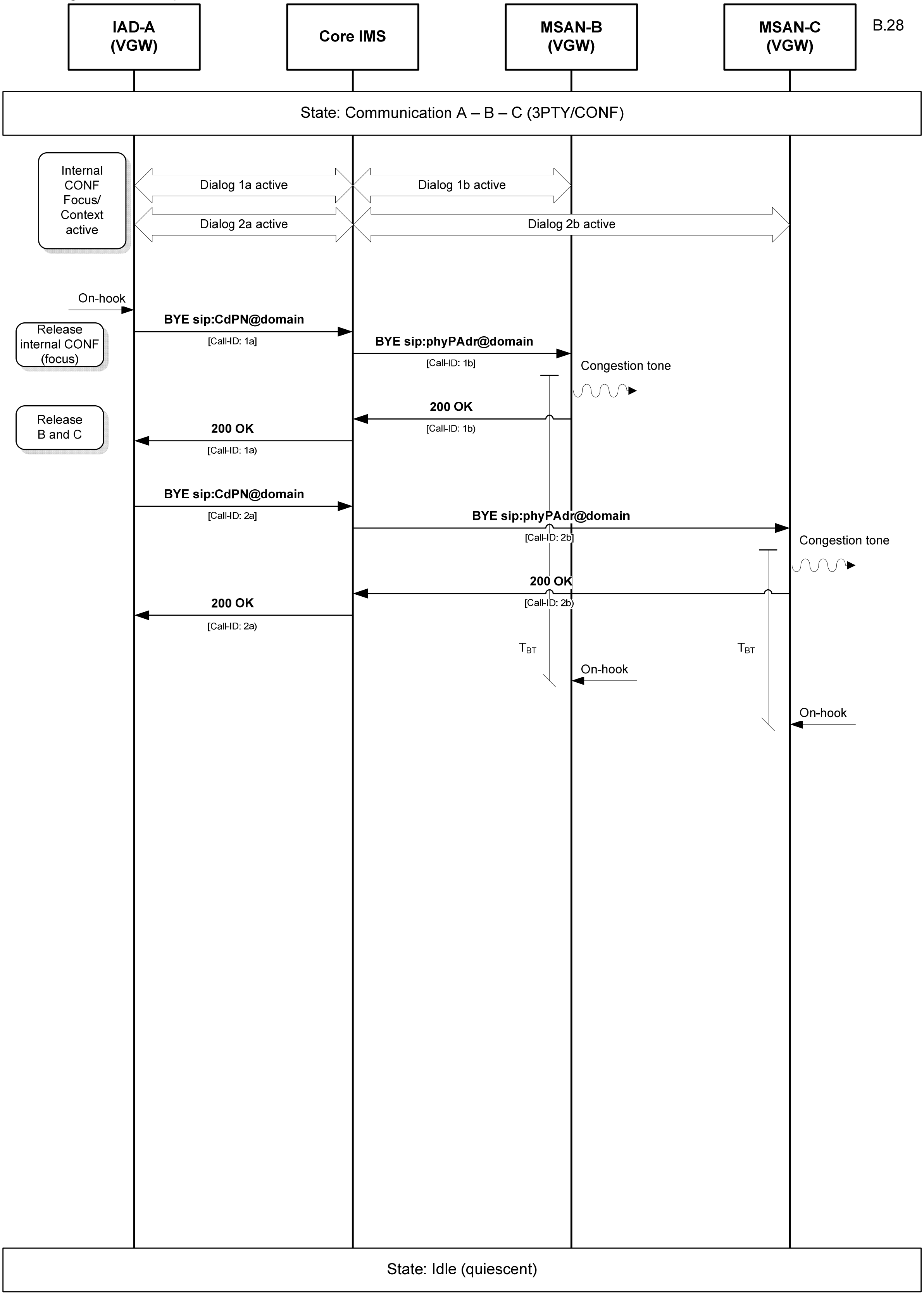
Conf Bridge must be implemented within the



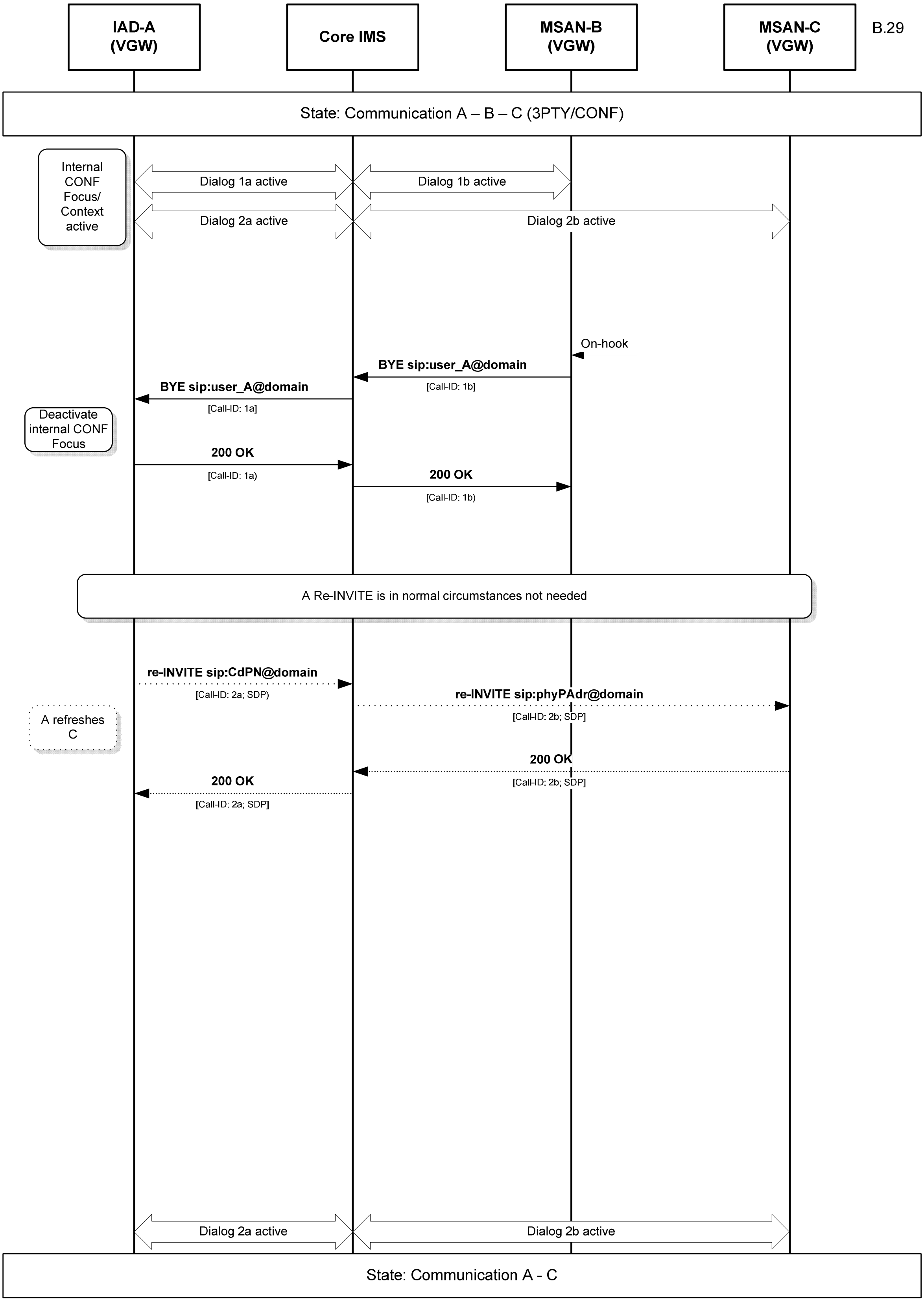
VGW.

B.28 3PTY: Release 3PTY/CONF by user A

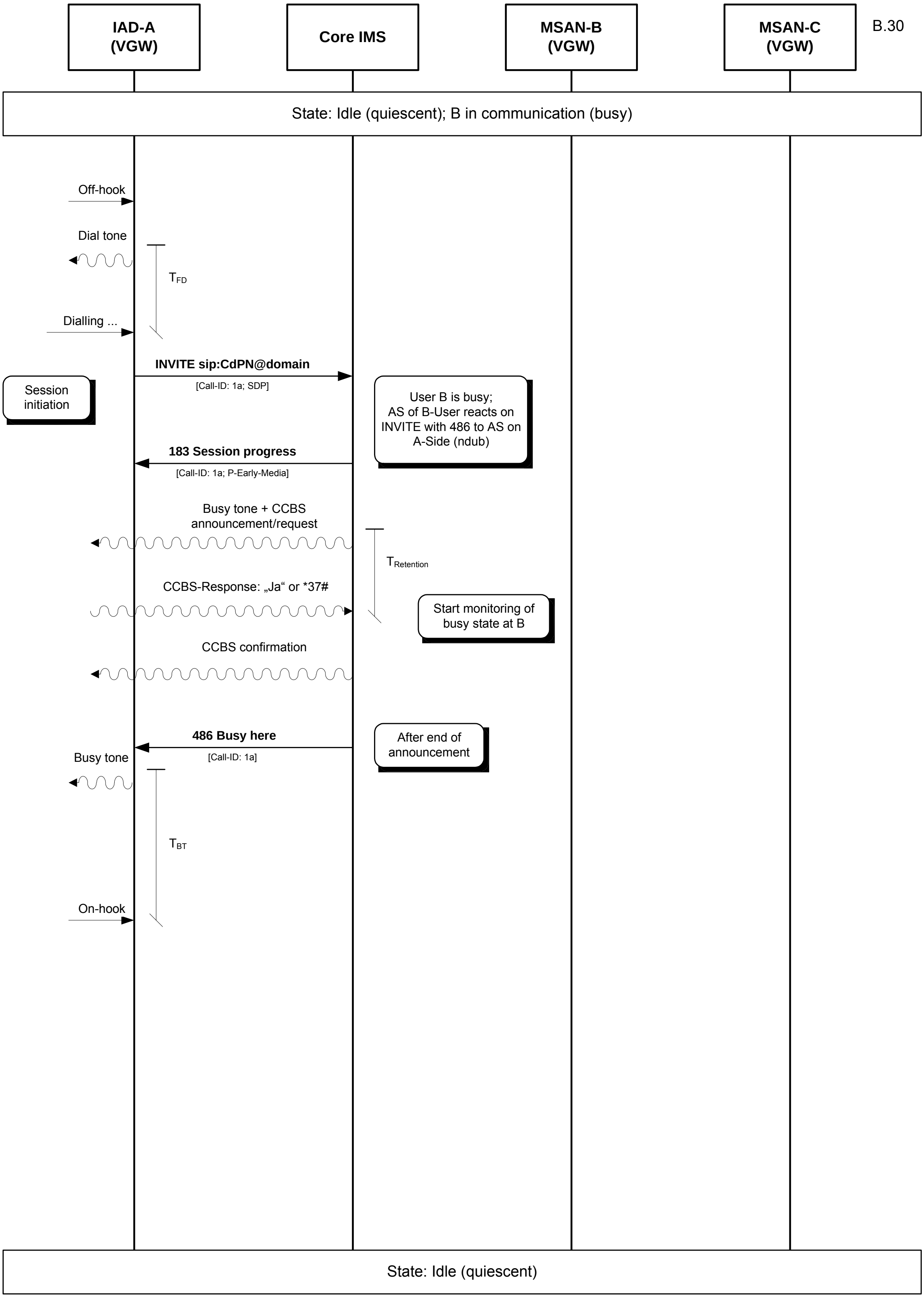
Conf Bridge must be implemented within the VGW.



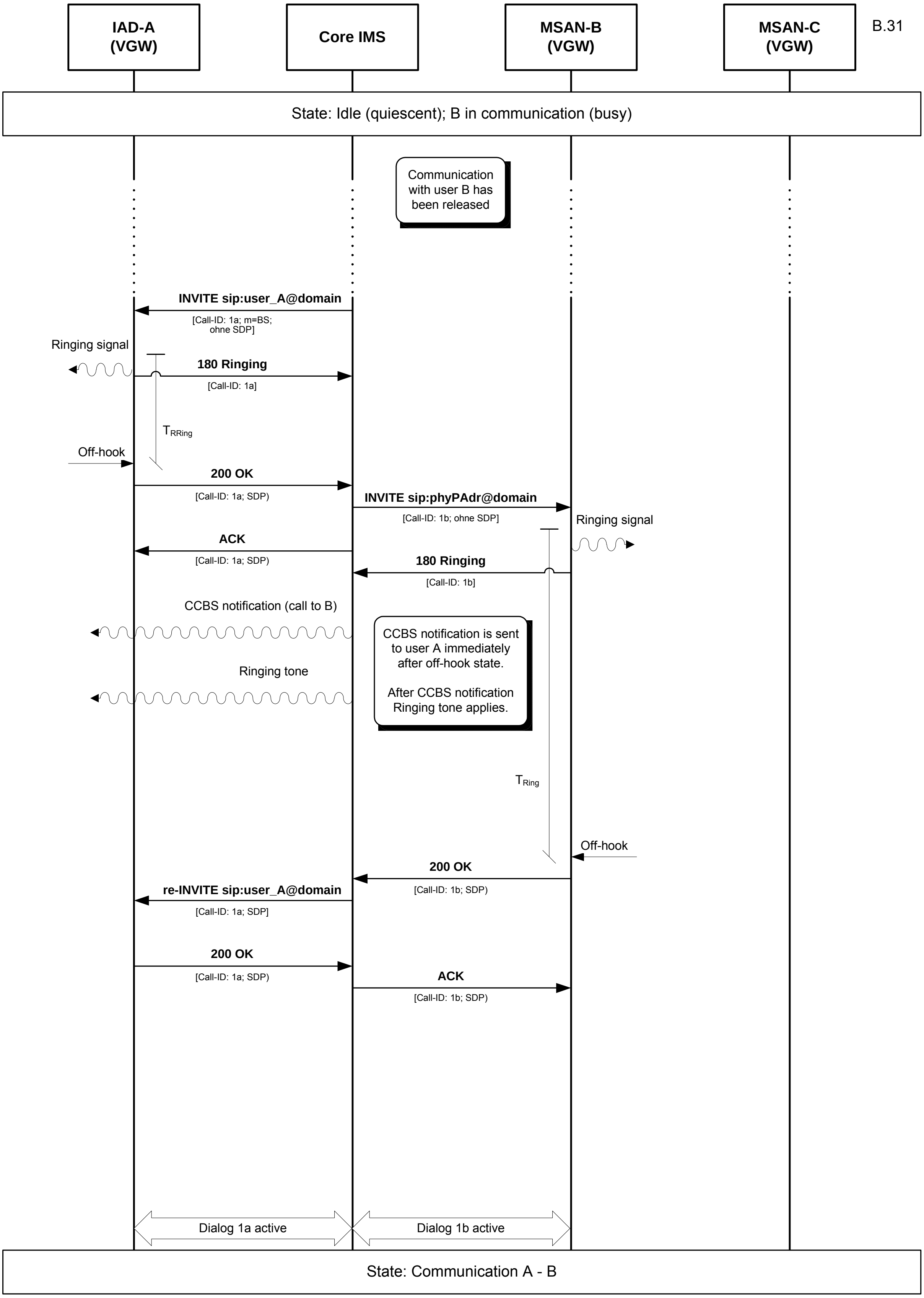
B.29 3PTY: Release 3PTY/CONF by user B



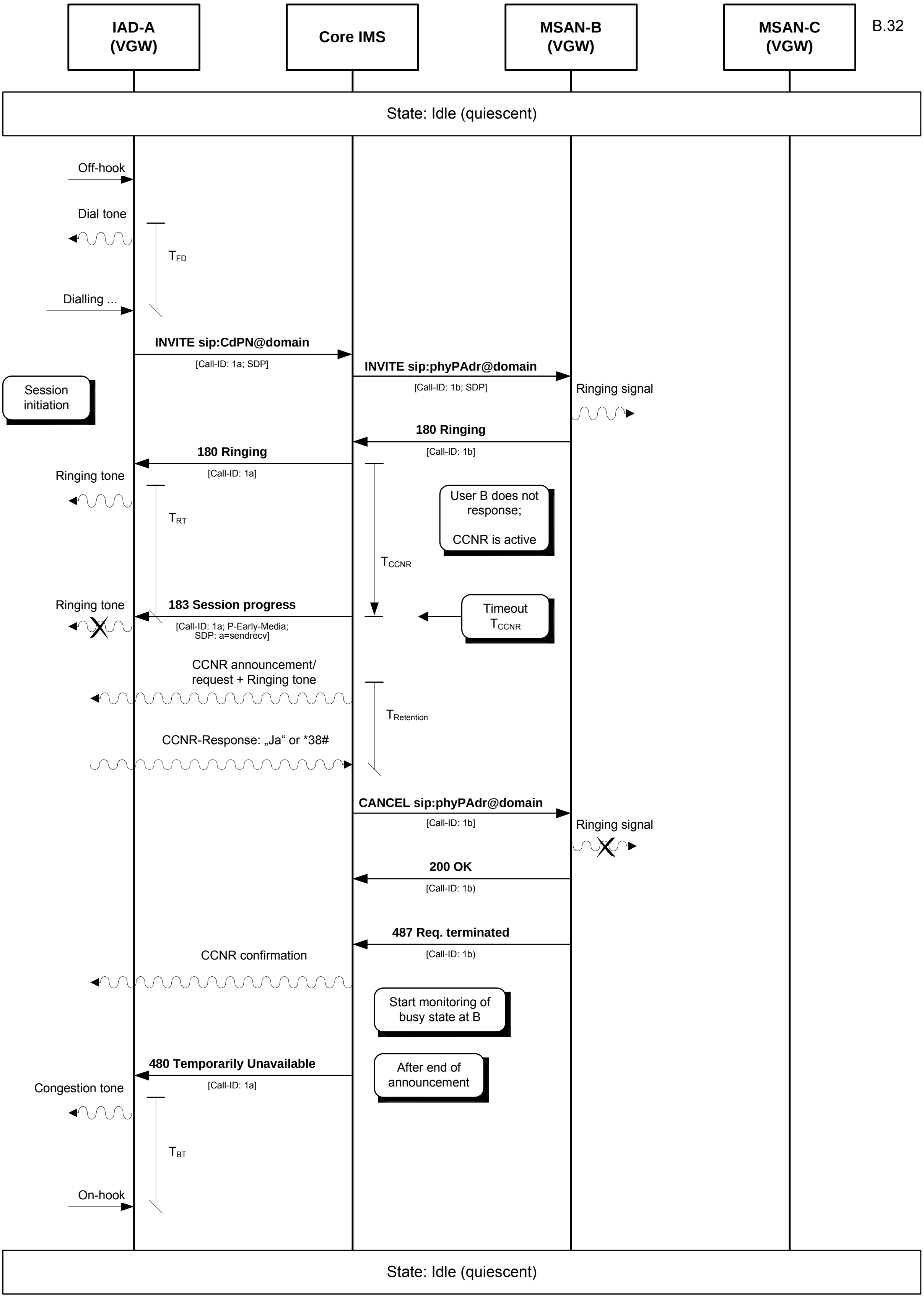
B.30 CCBS: Communication Completion on Busy Subscriber – Activation



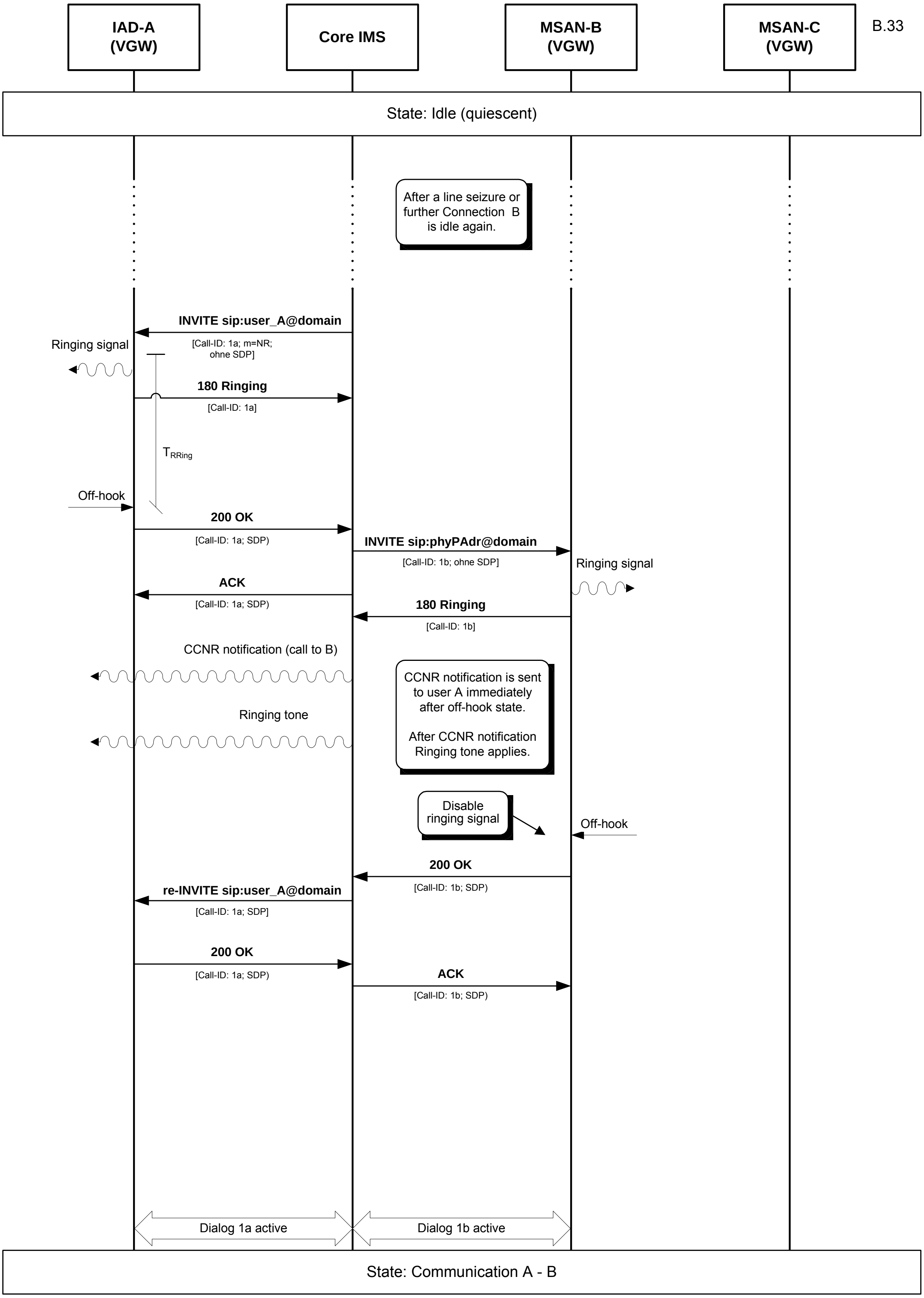
B.31 CCBS: Communication Completion on Busy Subscriber – Processing



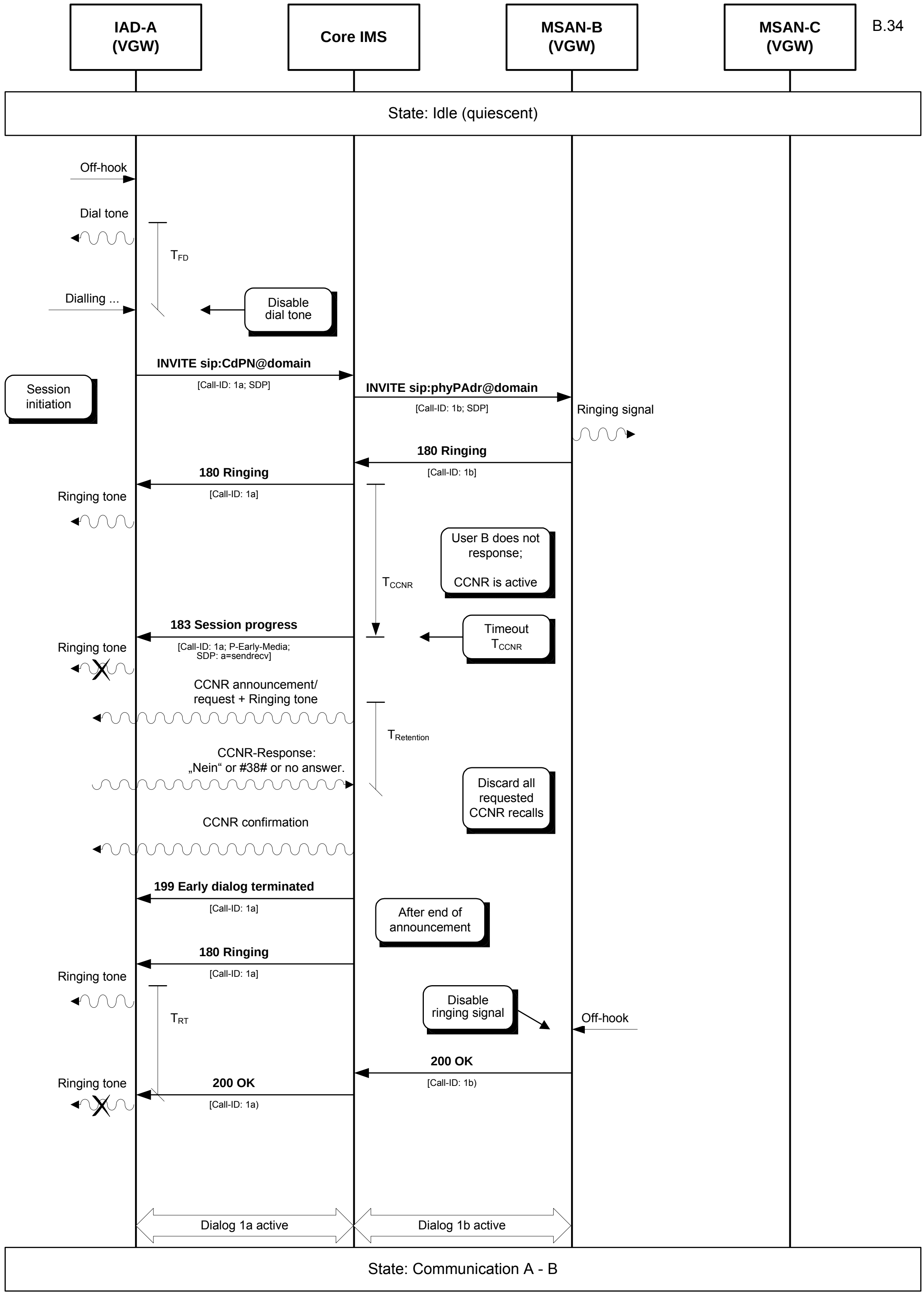
B.32 CCNR: Communication Completion on No Reply – Activation



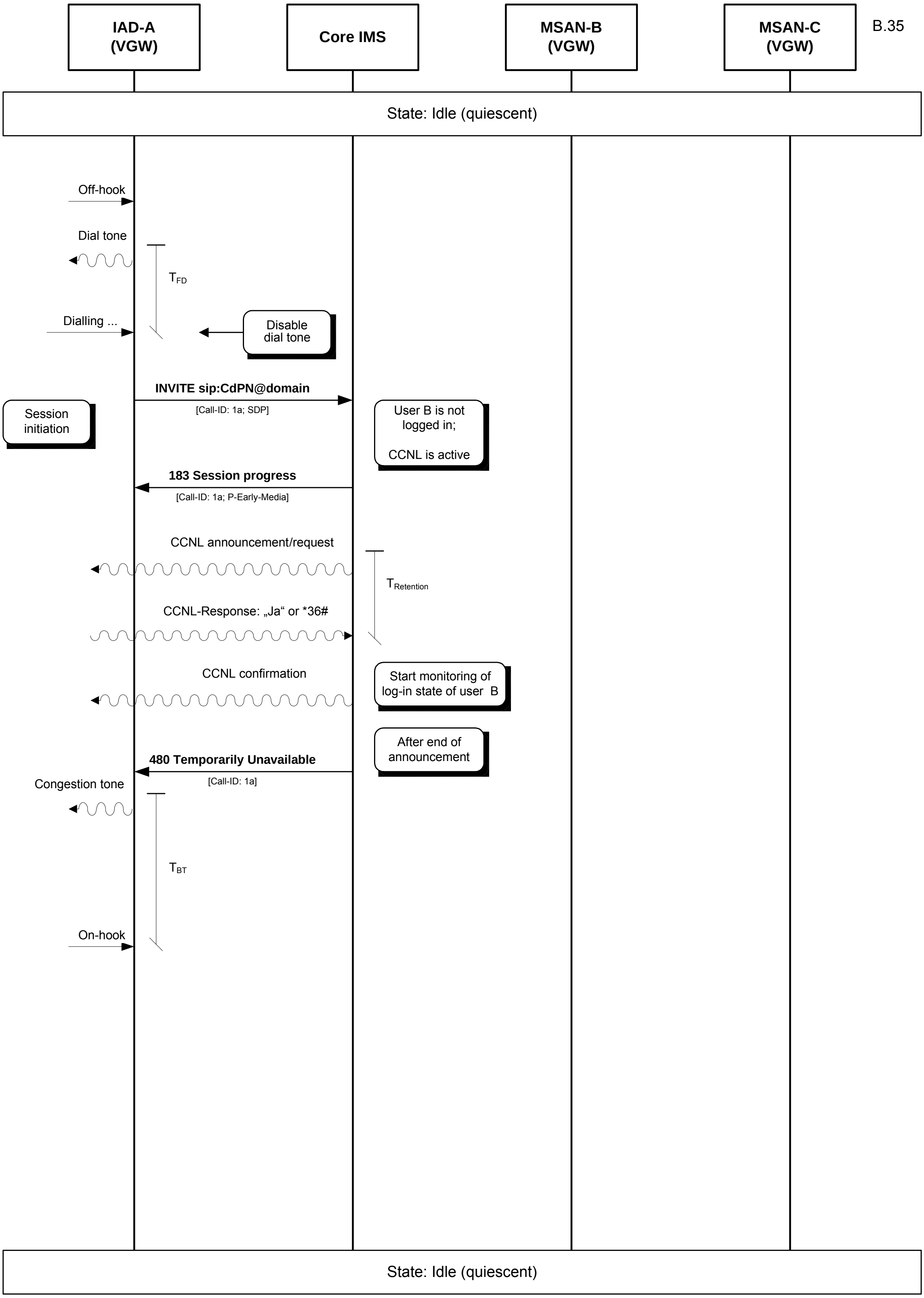
B.33 CCNR: Communication Completion on No Reply – Processing



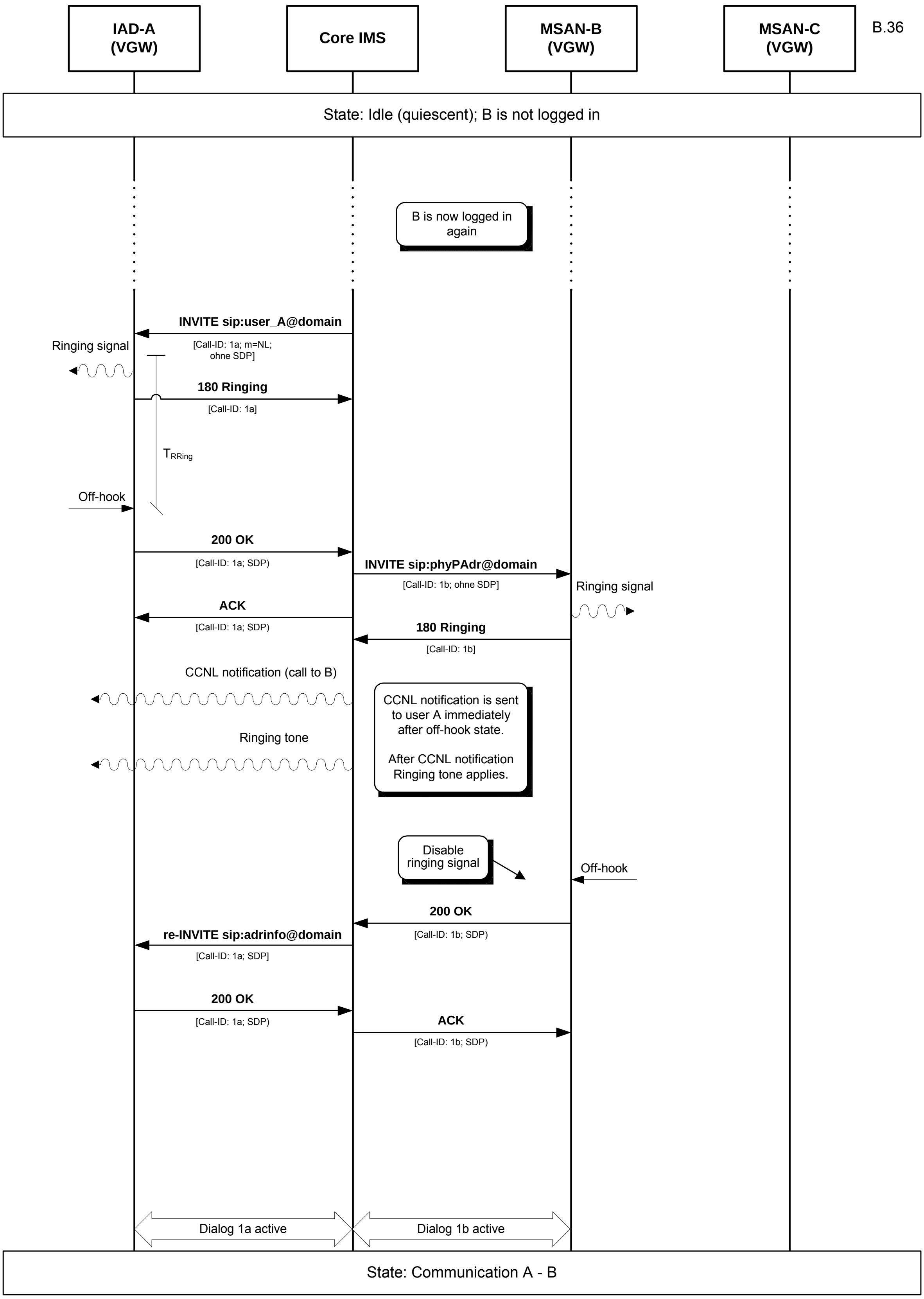
B.34 CCNR: Communication Completion on No Reply – Rejection



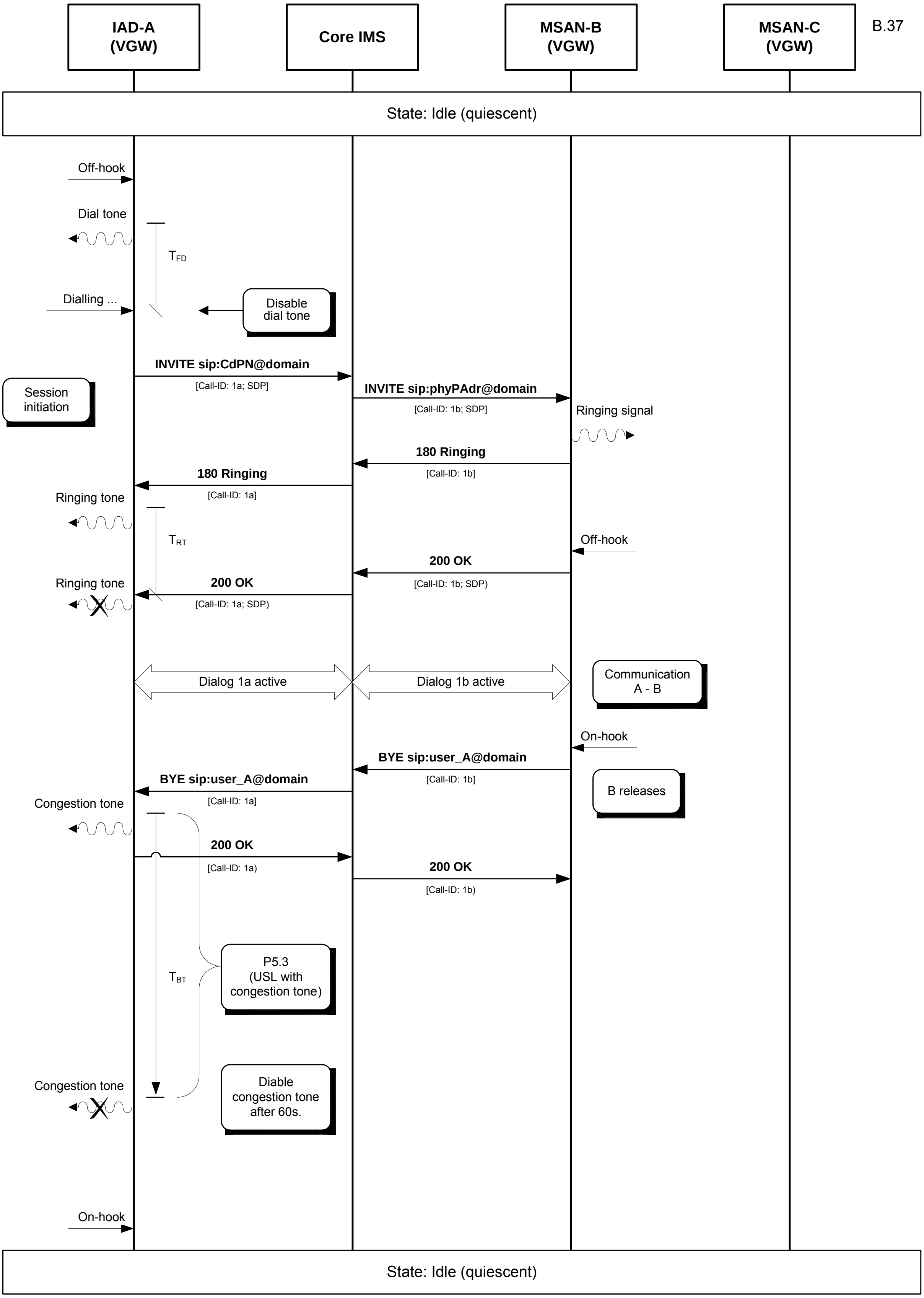
B.35 CCNL: Communication Completion Not Logged-in – Activation



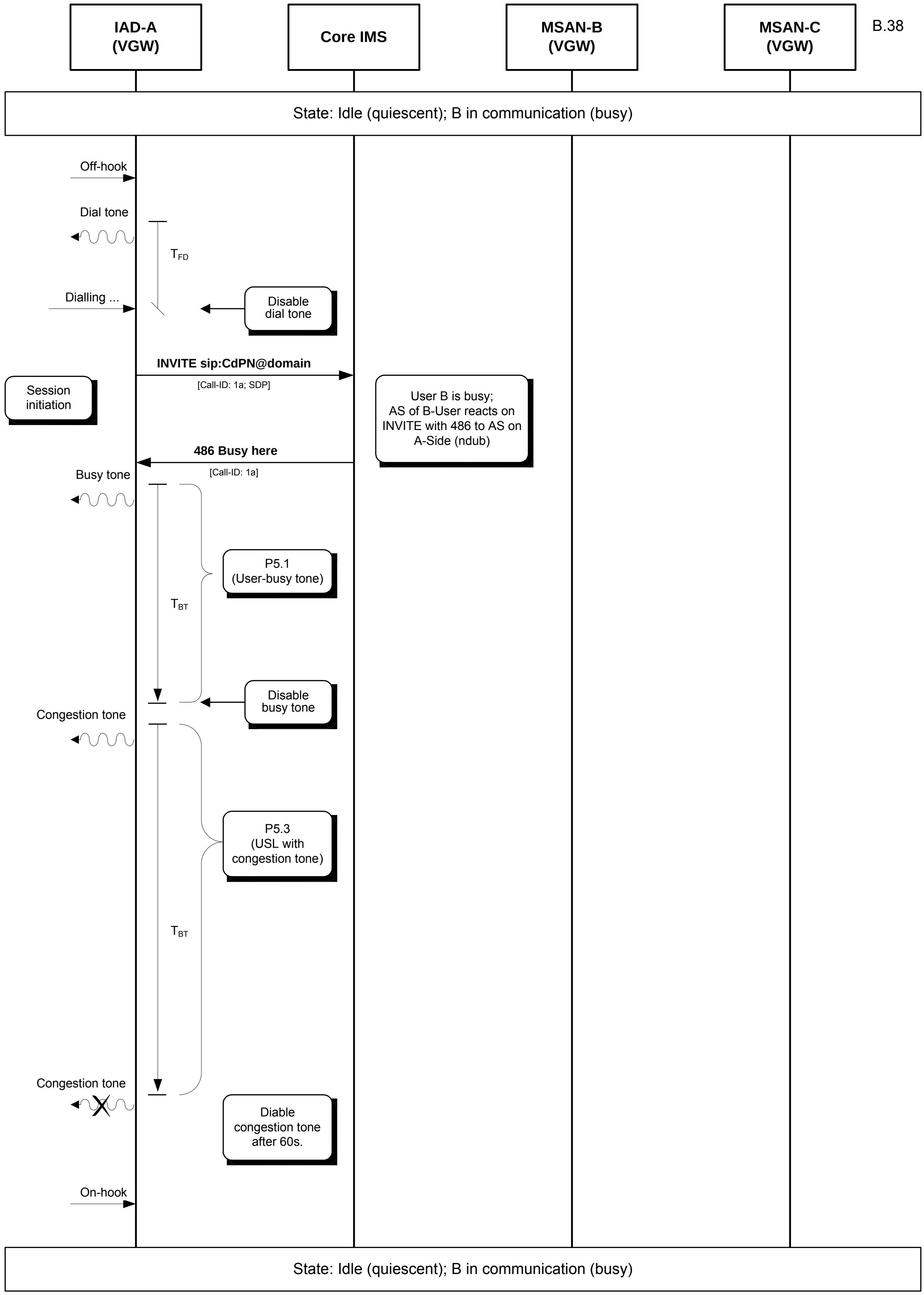
B.36 CCNL: Communication Completion Not Logged-in – Processing



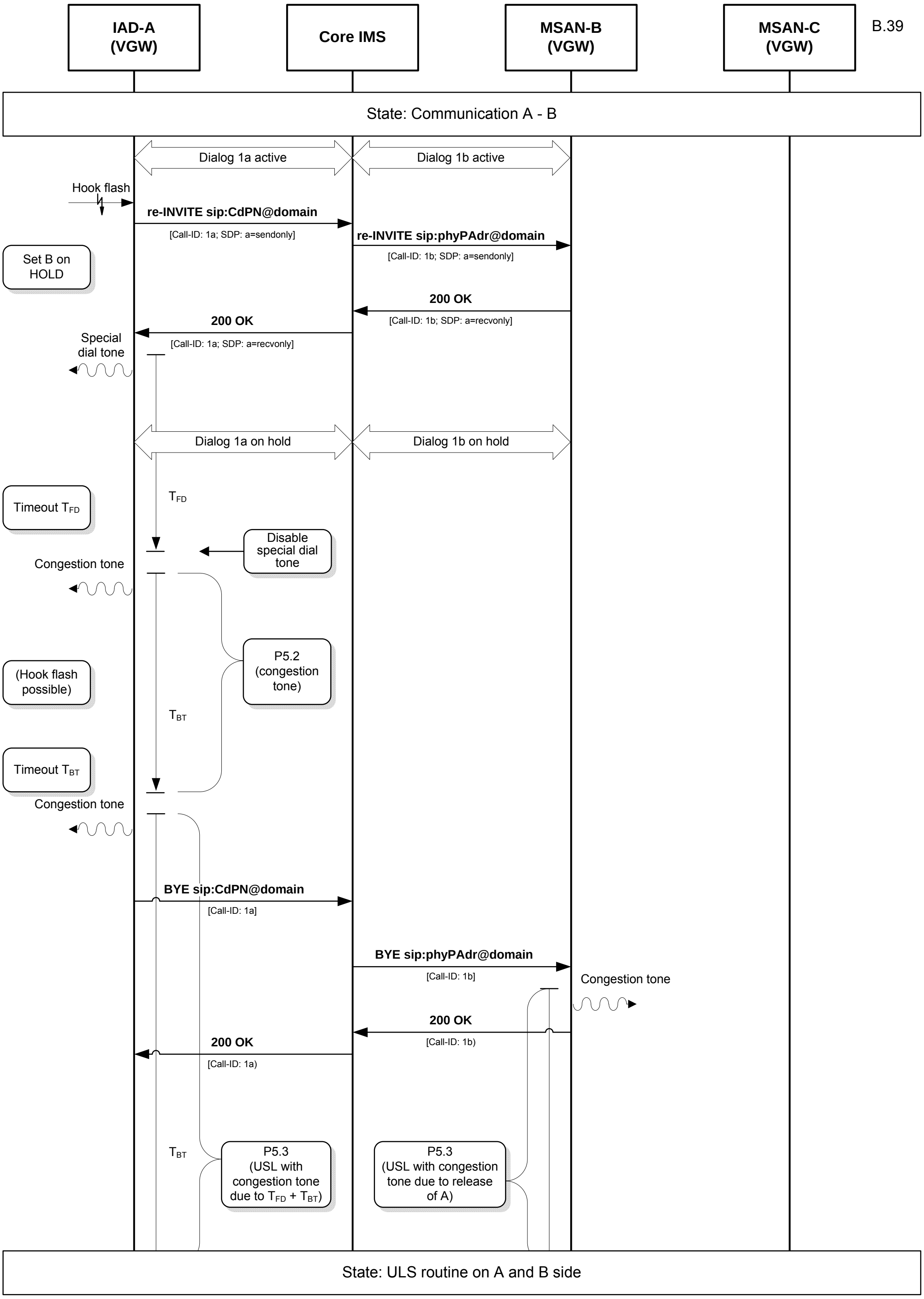
B.37 ULS after release of user B



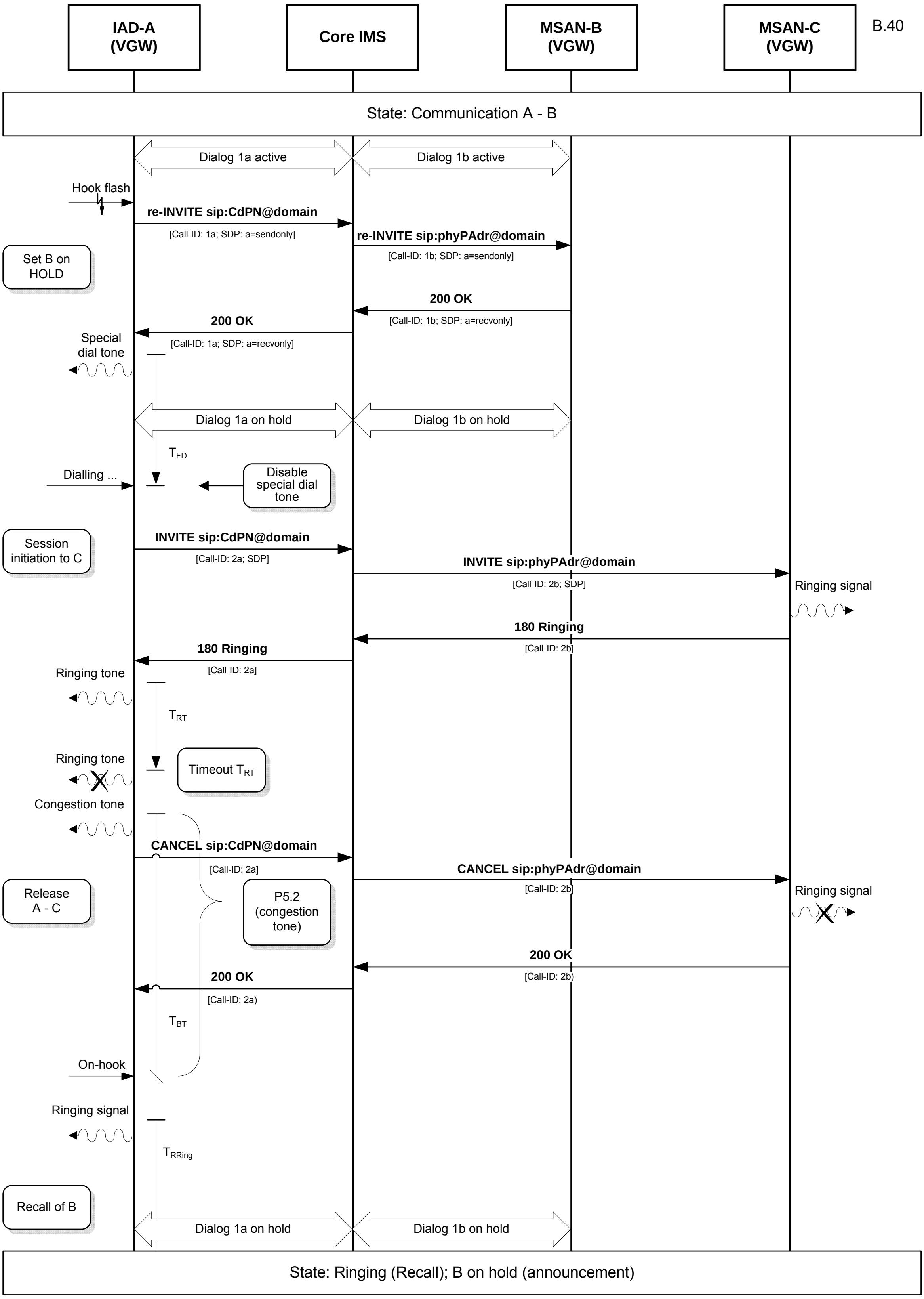
B.38 ULS after busy



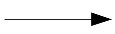
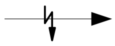
B.39 ULS after timer expiry of TFD



B.40 ULS: Resume communication before ULS



B.41 Legend

<table>
<tr><td rowspan="2"></td><td>a/b-Interface: (e.g. tones, announcements)</td></tr>
<tr><td>Network side: RTP-Streams</td></tr>
<tr><td rowspan="2"></td><td>a/b-Interface: analogue events (e.g. on-hook, digits)</td></tr>
<tr><td>Call Control side: SIP-Messages</td></tr>
<tr><td></td><td>Analogue event: Hook-flash</td></tr>
<tr><td></td><td>Timer start until timeout</td></tr>
<tr><td></td><td>Timer start until timer stop due to an event</td></tr>
<tr><td></td><td>Range of port-status</td></tr>
<tr><td></td><td>Not shown procedures</td></tr>
</table>

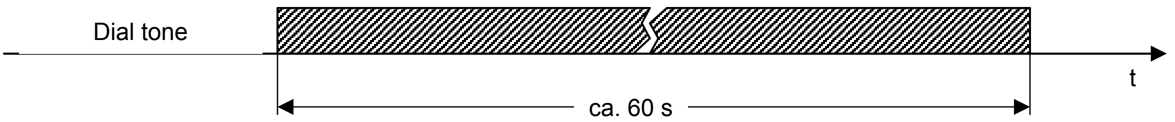
Annex C Signals and Supervisory tones

C.0 General

The levels for audible tones, 16 kHz metering pulses and FSK data transmission as well as the AC voltage for ringing signals shall be properly defined so that the required values for TEs according to 1TR110-1 [2] will be fulfilled. The maximum line length supported by the analogue ports of the VGW has to be considered, too. The following signals shall be provided by the VGW on each relevant analogue port. Additional signals are permitted for internal functions.

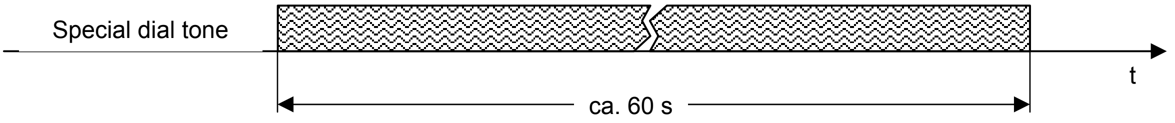
C.1 Dial tone

The dial tone is provided after line seizure (off-hook state) until the first dialling digit has been received and serves as information “ready for dialling” for the subscriber or specific terminal devices.
The dial tone shall be a permanent tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of $k \leq 5$ % as depicted in the figure below.



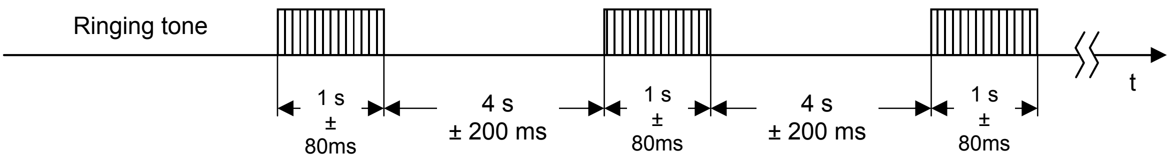
C.2 Special dial tone

The special dial tone is provided after line seizure (off-hook state) in conjunction with a particular case until the first dialling digit has been received and serves as information “ready for dialling” for the subscriber or specific terminal devices. In addition, the special dial tone gives the subscriber a hint about a specific situation (e.g. another party is on hold).
The special dial tone shall be a permanent tone with a frequency composition range of f_1 = 400 Hz ± 7 Hz and f_2 = 425 Hz ± 7 Hz (additive); the distortion factor shall be $k \leq 5$ % as depicted in the figure below.



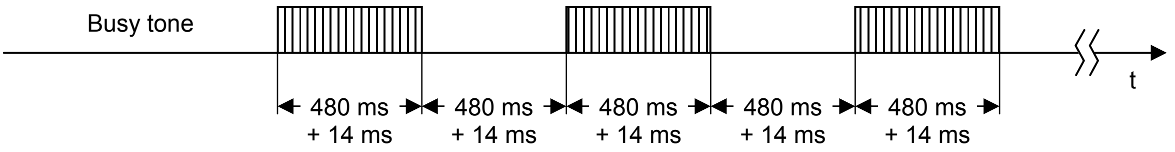
C.3 Ringing tone

The ringing tone is provided after a successful communication set-up is finalized and the destination party is idle. The ringing tone serves as information “ringing signal is provided to the destination party” for the subscriber or specific terminal devices.
The ringing tone shall be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.



C.4 Busy tone

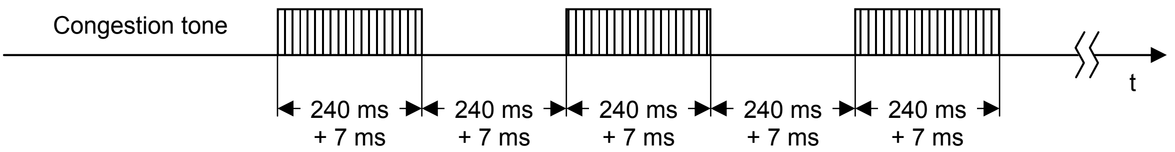
The busy tone is provided after a successful communication set-up is finalized and the destination party is busy. The busy tone serves as information “destination party busy” for the subscriber or specific terminal devices.
The busy tone shall be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.



C.5 Congestion tone

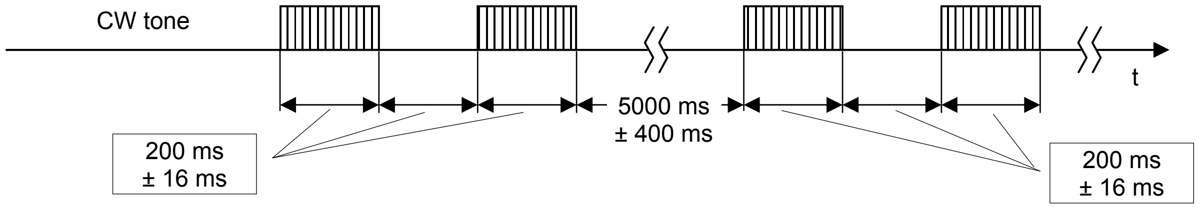
The congestion tone is provided in case of an unsuccessful communication set-up or in case of an error. The congestion tone serves as information “unsuccessful communication” or “operating error” (see bullet points below) for the subscriber or specific terminal devices.
The congestion tone shall be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.
The congestion tone is played in case the following causes:

- Communication set-up to the destination party is not possible,
- a time-out has expired,
- a garbage has been detected (e.g. SOC/SCC).



C.6 Communication waiting tone

The communication waiting tone (CW tone) is provided during the communication state while a third party communication is waiting. The CW tone serves as information "another communication is waiting" for the subscriber or specific terminal devices.
The communication waiting tone shall be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.



C.7 Metering pulse (16-kHz)

The 16-kHz metering pulse (tariff information) is provided during the communication state and shall be transmitted as an AC signal with a frequency range of 16-kHz ± 80 Hz and a distortion factor of k ≤ 8 %. The pulse duration shall be within the range of t_{pulse} = 110 ms ± 10 ms and the minimum pause between two pulses shall be t_{pause} ≥ 135 ms.

C.8 FSK data transmission

For the implementation of display services (e.g. OIP: CLIP on-hook/off-hook) the requirements according to ETSI standards EN 300 659, Part 1 [9] and Part 2 [10] as well as ES 200 659-3 [11] shall be fulfilled. Further details about the used options and parameters are described in 1TR110-1 [2] (section 9).

NOTE: If the CgPN includes a "+" sign, it shall be mapped to "00" in the CLI-parameter.

C.9 Ringing signal

The ringing signal for incoming calls (external call) is provided during the idle state and shall be provided as an AC signal with a frequency range of 25 Hz ± 2 Hz or optionally of 50 Hz ± 4 Hz and a distortion factor of k ≤ 8 %. The following ringing cadences shall be fulfilled (for external calls):

Table C.9-1

	Duration in ms				
1. ringing signal (pulse)	400	≤	500	≤	700
further ringing signals (pulse)	920	≤	1000	≤	1080
1. ringing signal pause	0	≤	5000	≤	5400
further ringing signal pauses	4600	≤	5000	≤	5400

Additional deviated ringing cadences for internal calls or features are possible.

History

Version	Published	Remarks
1.0	15.05.2009	First version of 1TR126.
2.0	31.07.2009	Second version of 1TR126 (English version); - Main body translated in English, - Editorial modifications, - Clause 3 updated and completed.
2.1	18.12.2009	Updated version of 1TR126: - Editorial modifications, - Version of reference to ETSI TS 183 043 [6] changed (V2.2.2 => ETSI interim draft version !), - Reference [10] updated from RFC 2833 to RFC 4733, - Clause 3.3 updated, - Timer definitions updated (inter-digit timer, Ta1, etc.), - Clause 4.2 updated due to general changes (e.g. no overlap sending, timer values, etc.), - Clause 4.2.9.3 and 4.2.9.4 added, - Clause 4.3 tables (port states and timers) updated, - AoC (reference and contraints changed: new => ETSI TS 124 447); - Annex A modified: - P-Preferred-Id as an Option, - Overlap signalling (In-Dialog) deleted, - Mapping of “+” into 00, - Note: Feature Codes, - Annex B updated and 2 new call flows (B.18 and B.25) added; - Annex C updated (C.8).
3.0	30.06.2010	Third edition of 1 TR 126: - Reference [8]; Version updated; - Old Annex A deleted (endorsement); - Modifications to ETSI TS 183 036 transferred into a new Annex A; - General (formal) modifications; - Clause 4.1: Echo requirements added; - Clause 4.2.4.1 and 4.2.4.2 updated; - Clause 4.2.4.3 added; - Clause 4.4 added; - Table 4-6 and 4-7 updated; - Formal corrections in Annex B and titles updated; - Annex B, B.3 and B.4 : Ta1 added - Annex B, B.5 and B.22 modified; - Annex B.26: "Sess-ID" replaced by "Session-ID"; - Annex B: titles updated; - Document properties and History updated; - Zentrum Technik Einführung replaced by

Version	Published	Remarks
		Fixed Mobile Engineering Deutschland; - T-Home replaced by Deutsche Telekom.
3.1.0		- 3PTY made to local feature within VGW - Replacement of Anouncement played by the network for Call Hold, Call Wait, TOGGLE and 3PTY - 4.2.2.1 editorial correction for better understanding - 4.2.8.1 adopt VGW internal procedures - 4.2.8.2 Internal VGW CONF (Focus) procedures described - 4.2.8.3 Internal VGW CONF disabling → return to HOLD - 4.2.8.4 SOC interaction for internal handling and State P4.3a - 4.2.8.5 Internal CONF release procedures added - 4.2.8.6 Internal CONF release procedures added - 4.2.9.2 Internal failure handling Dialling of unallowable digits adopted - 4.2.9.3 Internal failure handling Dialling of unallowable digits adopted - 4.2.9.4 Internal failure handling Dialling hang-up re-ring - Introduction of State P4.3a (SOC Interaction) due to gap and failure identified by integration. - Replacement of Call Flows B.6 – B.8 ANNEX B add additional text for internal handling of Call Wait, Call HOLD, 3PTY and TOGGLE B.4, B.5, B7 – B29, B39, B40, correction due to changes on internal handling of Call Wait, Call HOLD, 3PTY and TOGGLE