TTC STANDARDS

JJ-22.02

Inter-work Specifications between Private SIP Network and private ISDN Network

Version 1.2

June 9, 2016

THE TELECOMMUNICATION TECHNOLOGY COMMITTEE

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<Reference>

1. Introduction

The Private Network Interface Sub-Working Group of the Private Network Special Committee has implemented the standardization of the IP protocol based on private networks (line switching networks) between PBXs (Private Branch eXchanges) and Qsig (Signalling information flows at the Q reference point). Considering recent trends in markets and international recommendations, it is necessary to study VoIP (Voice over Internet Protocol) technology based on SIP (Session Initiation Protocol) within private networks. It has been decided to implement standardization by focusing on the latest technical trends in the new technical field mentioned above and the status of the responses of carriers to them.

This standard provides inter-work specifications between private SIP network and private ISDN (Integrated Services Digital Network) (Qsig) network, using as reference the SIP-ISUP (Integrated services digital network User Part) inter-work on carrier networks, with consideration given to response value conversion and multi-vendor support to be provided when connections are made to individual GWs (GateWays).

2. Revision History

Version	Date of establishment	Description
First Version	August 24, 2006	Established.
Version 1.1	December 6.2007	Revision.
Version 1.2	June 9,2016	Revision (correction of Figure 2-1)

3. Miscellaneous

(1) Recommendations, standards, etc., referenced

TTC standard : JF-IETF-RFC3398 Technical Specification on SIP to ISUP Interworking First Edition, June 2, 2005

JS-11572 : Private Integrated Services Network (circuit-mode bearer services) - Layer 3 Specifications for inter-PBX signalling protocol Third Edition, April 23, 2003

(2) Associations with other domestic standards

No associations with other domestic standards.

4. Organizational Unit Preparing Standards

First Version: Private Network Special Committee

Version 1.1: Private Network Special Committee

Version 1.2: Enterprise Network Working Group

1. General Aspect

This standard specifies the items that require special consideration in signalling inter-work between SIP and TTC-compliant JS-11572.

1.1. Contents of this standard

The contents of this standard are as follows:

This standard corresponds to (is similar to) JF-IETF-RFC3398, approved by TTC. The public network interface and the private network interface, however, differ in reference point (location), so that the provisions of JF-IETF-RFC3398 are not necessarily applicable to the private network interface. Thus, JS-11572 (Third Edition, recommendations established in April 23, 2003) is used to specify the specification conditions for inter-working to SIP (RFC3261), thereby assuring multi-vendor support.

1.2. Clarification of specification conditions

1.2.1. Range of the conditions of this standard

- In this standard, the scope of bearers is limited to voice or 3.1-kHz audio calls.
- In this standard, various additional service calls are outside the scope of study.

⇒ Transfer additional service, call completion additional service A

(Some calling line identification presentation and called number presentation services are covered.)

- In this standard, only interconnections between PISN (Private Integrated Services Network) and private SIP network are covered, and connections among PISN, SIP network, and PISN are not covered.

⇒ The PISN may employ channel associated signalling and, therefore, is covered in notes on a separate sheet.

1.2.2. Ranges that particularly need to be clarified

- Presentation of the information element in each message field

The capability of each information element shall conform to the range of its field in JS-11572.

Table 1-1/JJ-22.02 Presentation of the information element of each message field

Information element	JS-11572	SIP	Whether can be presented
Called number	Called number information element	SIP-URI	Yes
Calling line identification	Calling line identification information element	From	Yes
Name information	NAME-ID information element	From	Yes
Low/high layer	Low/high layer	None	No

Reason value	Cause information element	Response value	Partially not convertible

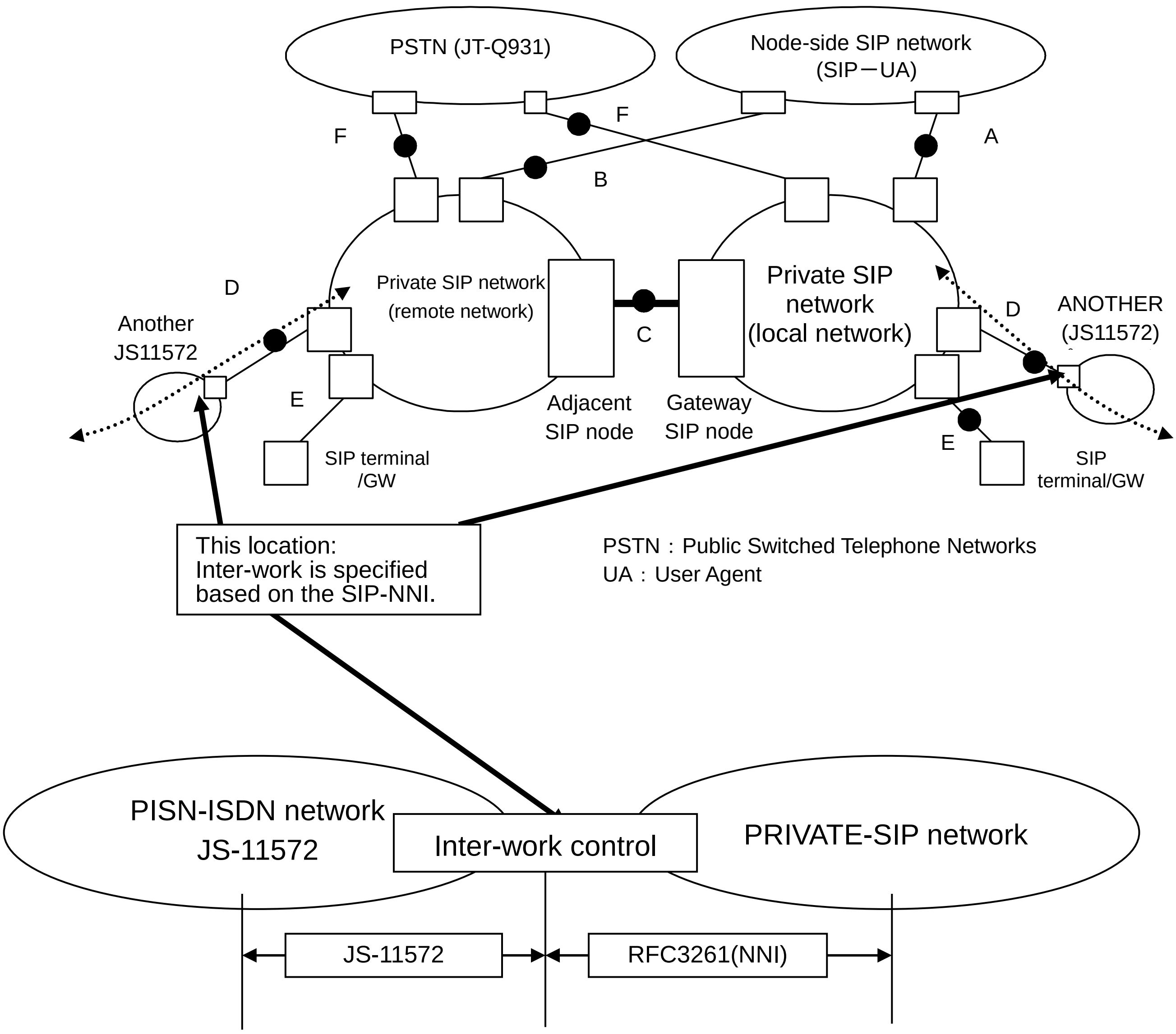
- Association between JS-11572 and SIP messages

  Signal sequence during inter-work: Survey/study, including tone control

- Method to set a calling line identification and the interworking indicator when mapping a SIP INVITE to a JS-11572 SETUP message, and the handling of the called number field
- Assurance of affinity with each protocol timer when SIP and JS-11572 inter-work with each other

2. Connection Reference Points

Regarding connection reference points, a protocol is specified by focusing on the inter-work of the Q reference point with the SIP-NNI (Network-Network Interface), as shown below. The reason for the definition with the inter-work with the SIP-NNI is to take into consideration inband control with the existing network and the affinity of information elements.



Configuration 1: Private ISDN-to-private SIP network connection

Figure 2-1 /JJ-22.02 SIP connection architecture

3. Presentation of the Information Element of Each Message Field

The protocol scheme conforming to this standard **must** have a mapping function for each signalling system in the inter-work control part.

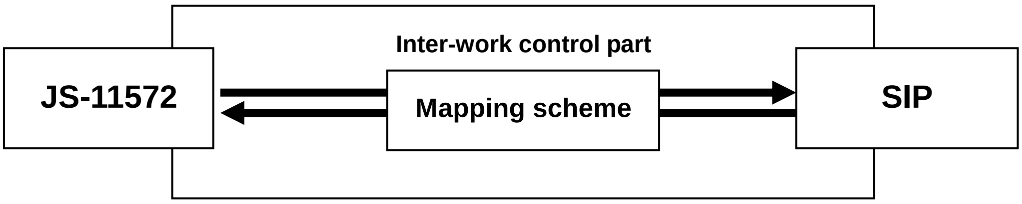


Figure 3-1 /JJ-22.02　Presentation of the information element of each message field

3.1. Called party number information presentation

- Incoming (JS-11572), outgoing (SIP)

If the inter-work control part is to perform routing with number dereferencing, it is an indispensable condition that the part uses the called party number information area from the ISDN to request a destination. If called party number information is to be carried on an INVITE, it is preferable that the same information be carried on the RQT (Request)-URI (Uniform Resource Identifier) and the TO header. The called party number may be converted by the inter-work control part.

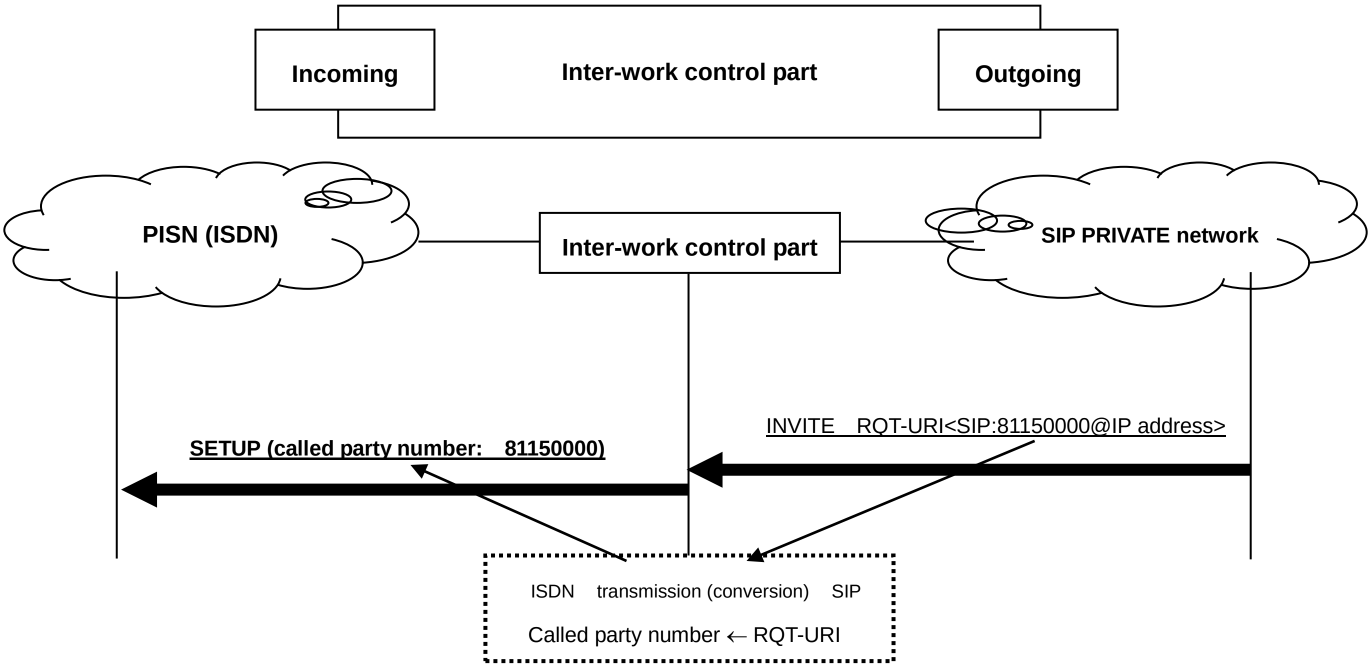


Figure 3-2 /JJ-22.02　Called party number information presentation (incoming (JS-11572), outgoing (SIP))

- Incoming (SIP), outgoing (JS-11572)

If the inter-work control part is to perform routing with number dereferencing, it is an indispensable condition that the part uses the request line to request a destination. Also, if SIP information is to transmit through the called party number of call establishment information (SETUP message), RQT-URI information shall be presented.

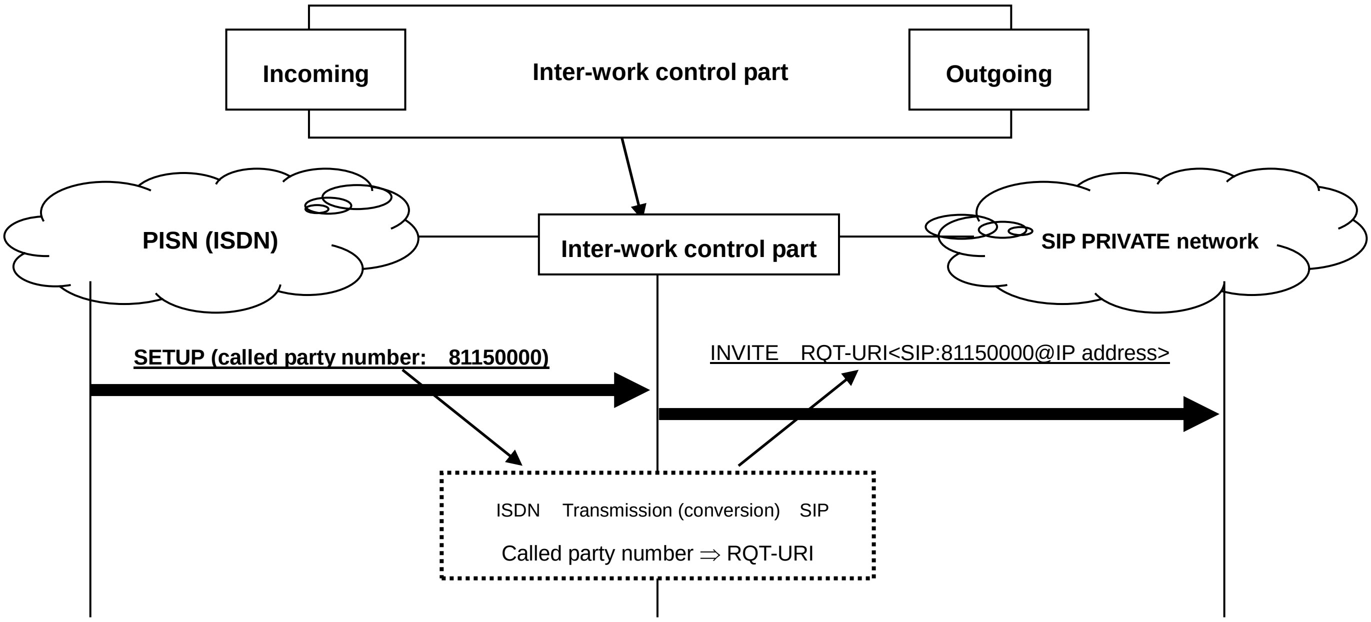


Figure 3-3 /JJ-22.02 Called party number information presentation (incoming (SIP), outgoing (JS-11572))

Supplementary information: This condition is also effective to connections to inter-work devices having ISDN user/network interfaces and to non-SIP signalling systems.

3.2. Handling of the mapping of calling line identification information

- Incoming (SIP), outgoing (JS-11572)

If the inter-work control part is to cause a calling line identification to transmit, it shall cause the username information in the From information in the INVITE to transmit through the calling party number information of call set-up information (SETUP message).

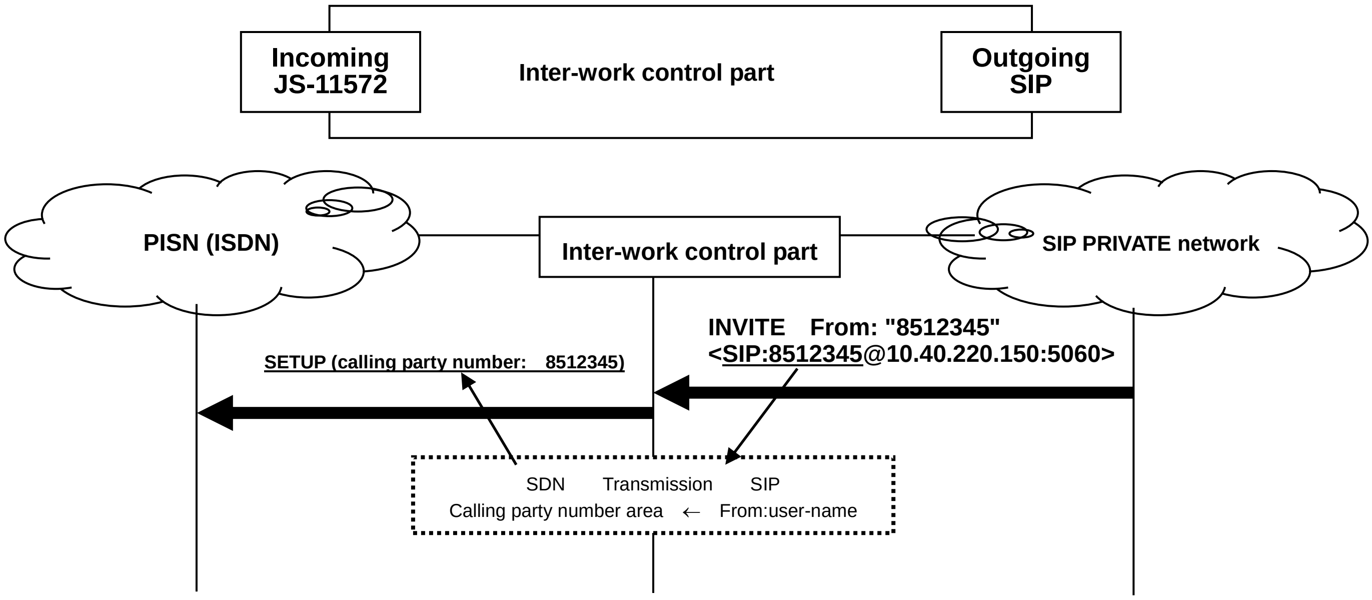


Figure 3-4 /JJ-22.02 Calling line identification information presentation (incoming (SIP), outgoing (JS-11572))

- Incoming (JS-11572), outgoing (SIP)

If the inter-work control part is to cause the calling party number to transmit, it shall cause it to transmit through the username information in the From information in the INVITE.

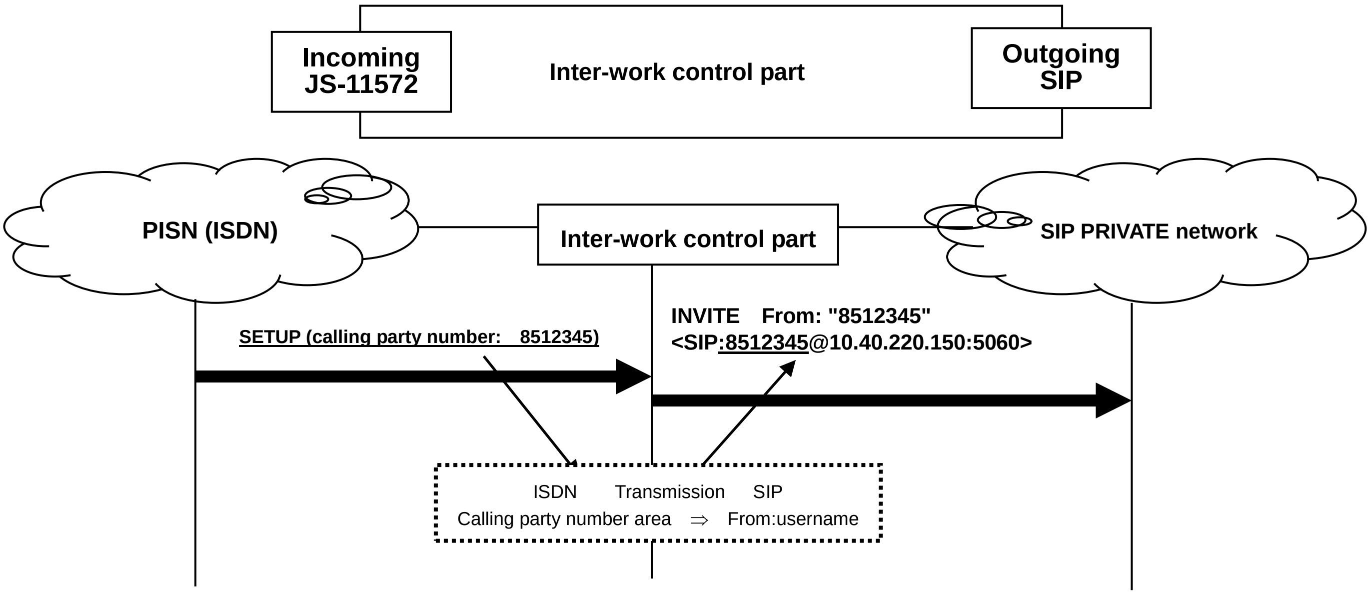


Figure 3-5 /JJ-22.02 Calling line identification information presentation (incoming (JS-11572), outgoing (SIP))

3.3. Mapping between ISDN <private network> TTC-JS-11572 and SIP response value

This section describes the positions of the cause value field and the response code field for the inter-work contorol part that are used to present the non-connection reason from the destination node/intermediate node to the originating node if a connection fails for some reason or other when inter-work is attempted for a call connection between JS-11572 and SIP. Based on the information below, the conversion rules for the inter-work control part are defined.

Supplementary information: Basically, the rules should be defined based on RFC3398, but if RFC3398 is cited, JS-11572 lacks some Reason values, and the rules are defined with consideration given to them.

In the case of non-conformance to this standard, the validity of mapping should be considered, using the reason header.

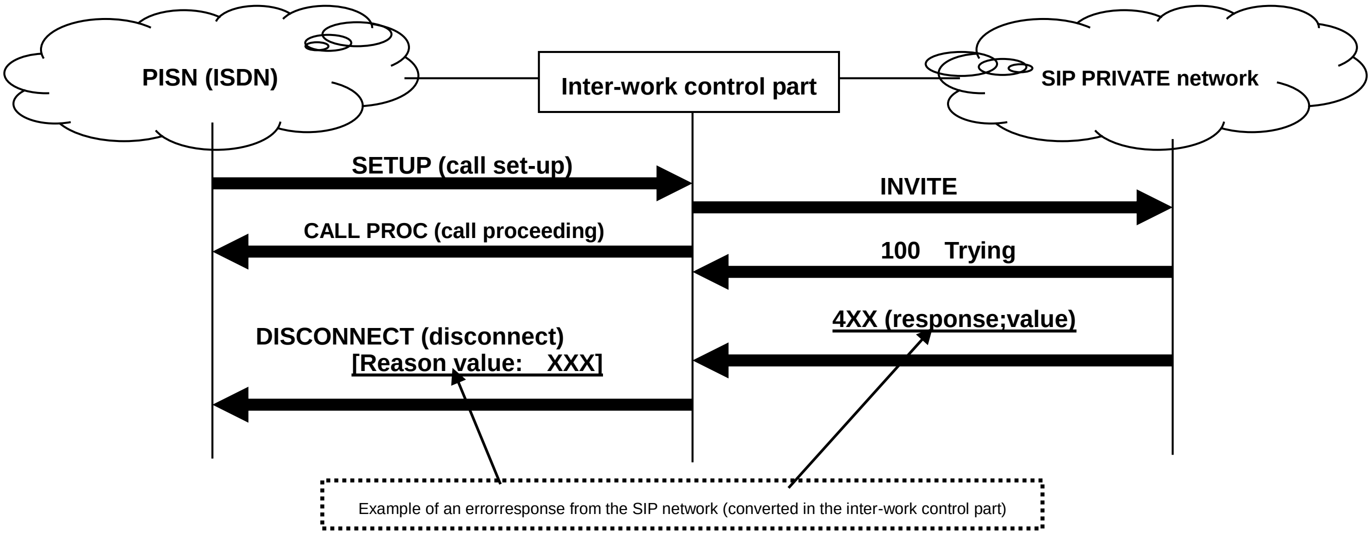


Figure 3-6 /JJ-22.02 Example of an error response from the SIP network

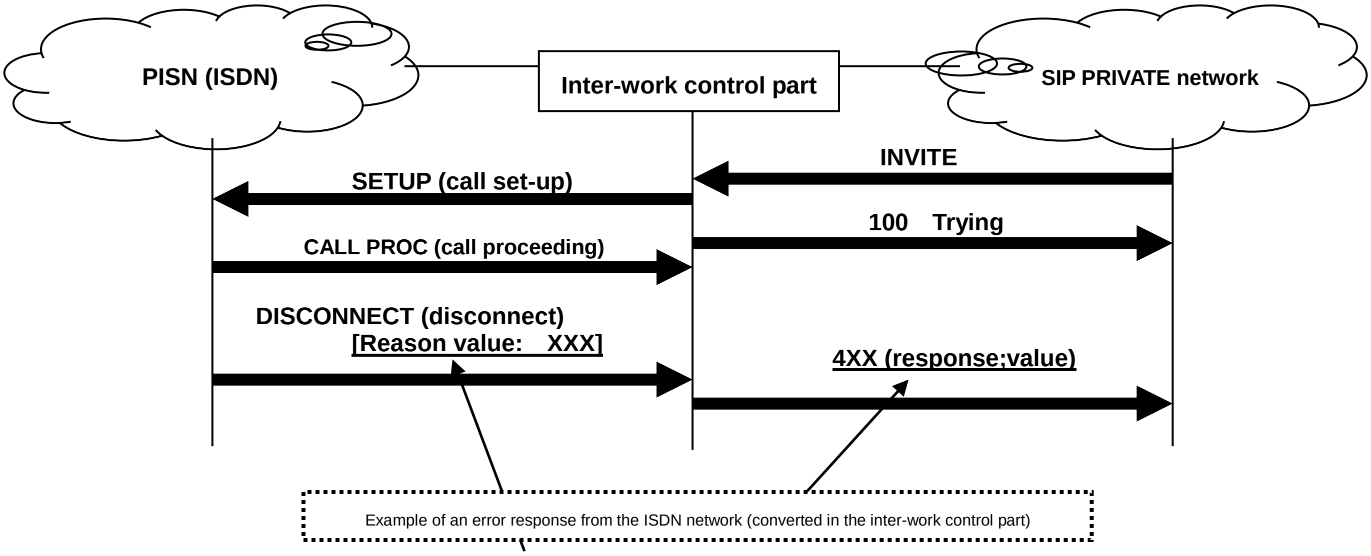


Figure 3-7 /JJ-22.02 Example of an error response from the ISDN network

3.3.1. Mapping between ISDN <private network> JS-11572 and SIP response value

Table 3-1 /JJ-22.02 Mapping between ISDN <private network> JS-11572 and SIP response value

Item No.	Mapping from JS-11572 (ISDN) to SIP		Difference from JF-IETF-RFC3398
	JS-11572 (ISDN)	SIP	
1	＃1 Unallocated (unassigned) number	404 Not found	
2	＃3 No route to destination	500 Server internal error	
3	＃6 Channel unacceptable	500 Server internal error	
4	＃16 Normal clearing	BYE/CANCEL (no response)	
5	＃17 User busy	486 Busy here	
6	＃18 No user responding	408 Request timeout	
7	＃19 User alerting, no answer	480 Temporarily unavailable	
8	＃21 Call reject	403 Forbidden	
9	＃22 Number changed	301 Moved permanently, if information in diagnostic field of Cause information element is suitable for generating a SIP Contact header. Otherwise: 410 Gone	
10	＃27 Destination out of order	502 Bad gateway	
11	＃28 Invalid number format (incomplete number)	484 Address incomplete	
12	＃30 Response to status enquiry	Not applicable	
13	＃31 Normal, unspecified	480 Temporarily unavailable	
14	＃34 No circuit/channel available	503 Service unavailable	
15	＃41 Temporary failure	503 Service unavailable	
16	＃44 Requested circuit or channel not available	503 Service unavailable	Not mentioned in 3398.

17	＃57 Bearer capability not authorized	403 Forbidden	
18	＃58 Bearer capability not presently authorized	503 Service unavailable	
19	＃63 Service or option not available, unspecified	500 Server internal error	
20	＃65 Bearer service not implemented	488 Not acceptable here	
36	＃81 Invalid call reference value	403 Forbidden	Not mentioned in 3398.
37	＃82 Identified channel does not exist	403 Forbidden	Not mentioned in 3398.
43	＃88 Incompatible destination	503 Service unavailable	
46	＃96 Mandatory information element is missing	403 Forbidden	Not mentioned in 3398.
47	＃97 Message type non-existent or not implemented	500 Server internal error	
48	＃98 Message not compatible with call state or message type non existent or not implemented	500 Server internal error	Not mentioned in 3398.
49	＃99 Information element non-existent	500 Server internal error	
50	＃100 Invalid information element contents	403 Forbidden	Not mentioned in 3398.
51	＃101 Message not compatible with call state	403 Forbidden	Not mentioned in 3398.
52	＃102 Recovery on timer expiry	504 GW timeout	
53	＃111 Protocol error, unspecified	500 Server internal error	

3.3.2. Mapping from ISDN <private network> SIP to JS-11572 response value

Table 3-2 /JJ-22.02 Mapping from ISDN <private network> SIP to JS-11572 response value (1/2)

Item No.	Mapping from SIP to JS-11572		Difference from JF-IETF-RFC3398
	SIP	ISDN	
1	400 Bad request	#41 Temporary failure	
2	401 Unauthorized	#21 Call reject	
3	402 Payment required	#21 Call reject	
4	404 Not found	#1 Unallocated (unassigned) number	
5	403 Forbidden	#21 Call reject	
6	405 Method not allowed	#63 Service or option not available, unspecified	
7	406 Not acceptable	#41 Temporary failure	#79 (not specified)
8	407 Proxy Authentication required	#21 Call reject	
9	408 Request timeout	#21 Call reject	
10	410 Gone	#22 Number changed	
11	413 Request entity too large	#21 Call reject	#127 (not specified)
12	414 Request-URI too long	#100 Invalid information element contents	#127 (not specified)
13	415 Unsupported media type	#41 Temporary failure	#79 (not specified)
14	416 Unsupported URI scheme	#100 Invalid information element contents	#127 (not specified)
15	420 Bad extension	#100 Invalid information element contents	#127 (not specified)
16	421 Extension required	#100 Invalid information element contents	#127 (not specified)
17	423 Interval too brief	#100 Invalid information element contents	#127 (not specified)
18	480 Temporarily unavailable	#18 No user responding	
19	481 Call/transaction does not exist	#41 Temporary failure	
20	482 Loop detected	#34 No circuit/channel available	
21	483 Too many hops	#63 Service or option not available, unspecified	#25 (not specified)
22	484 Address incomplete	#28 Invalid number format (incomplete number)	
23	485 Ambiguous	#1 Unallocated (unassigned) number	
24	486 Busy here	#17 User busy	
25	487 Request terminated	#31 Normal, unspecified, user busy	
26	488 Not Acceptable Here	#31 Normal, unspecified	
27	500 Server internal error	#41 Temporary failure	
28	501 Not implemented	#41 Temporary failure	#79 (not specified)
29	502 Bad gateway	#27 Network failure	#38 (not specified)
30	503 Service unavailable	#41 Temporary failure	

Table 3-2 /JJ-22.02 Mapping from ISDN <private network> SIP to JS-11572 response value (2/2)

31	504 Gateway timeout	＃102　Recovery on timer expiry	
32	505 Version not supported	＃ 63　Service or option not available, unspecified	#127 not specified
33	513 Message too large	＃ 63　Service or option not available, unspecified	#127 not specified
34	600 Busy everywhere	＃34　No circuit/channel available	
35	603 Decline	＃21　Call reject	
36	604 Does not exist anywhere	＃1　Unallocated (unassigned) number	
37	606 Not acceptable	＃65　Bearer capability not implemented	

3.4. Inter-work signal sequences

This section shows signal sequences for inter-work.

3.4.1. Normal sequence (transmission from SIP private network to PISN), no early media control, clearing by the called party

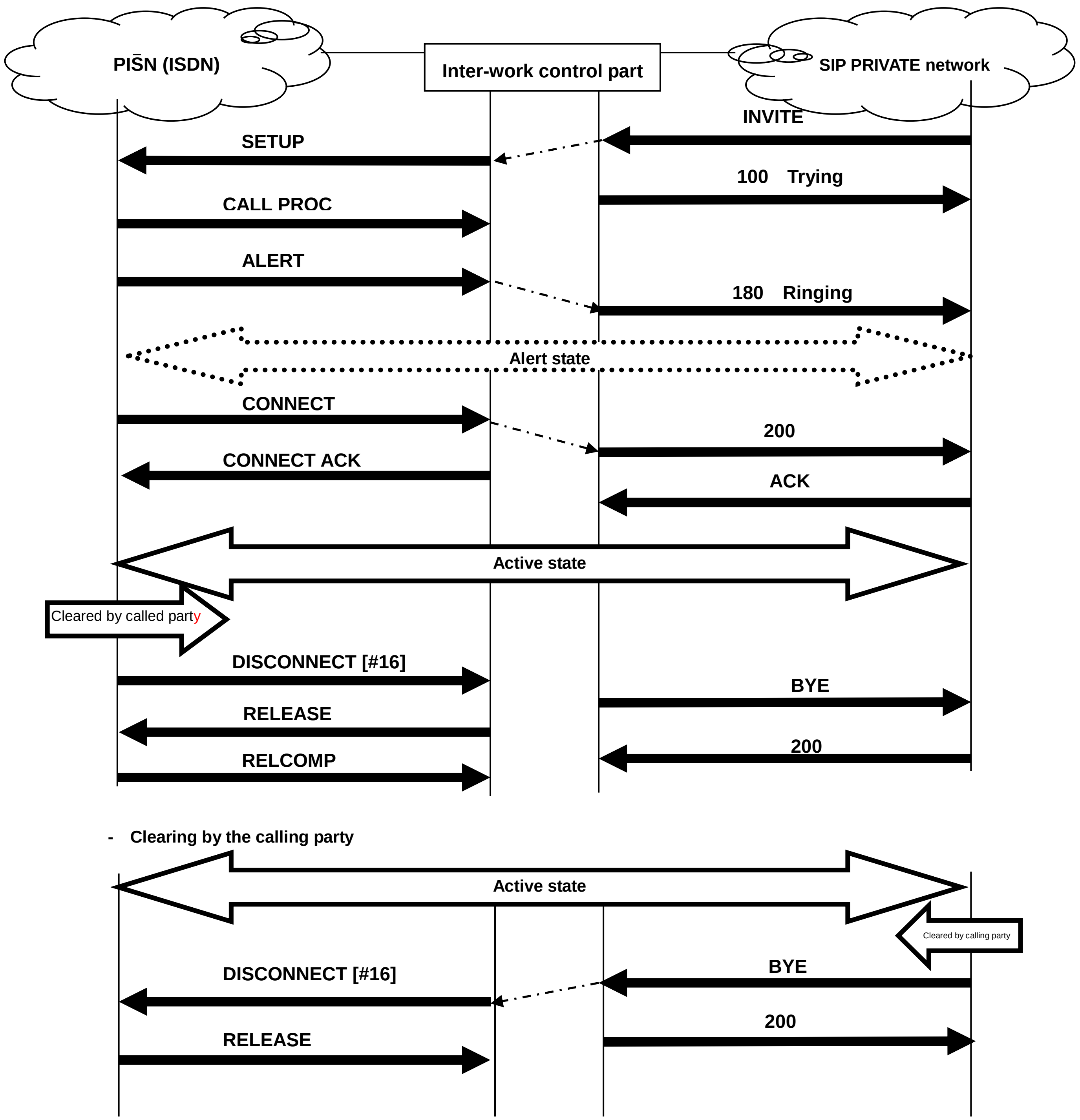


Figure 3-8/JJ-22.02 Normal sequence (transmission from SIP private network to PISN)

3.4.2. Normal sequence (transmission from PISN to SIP network), no early media control

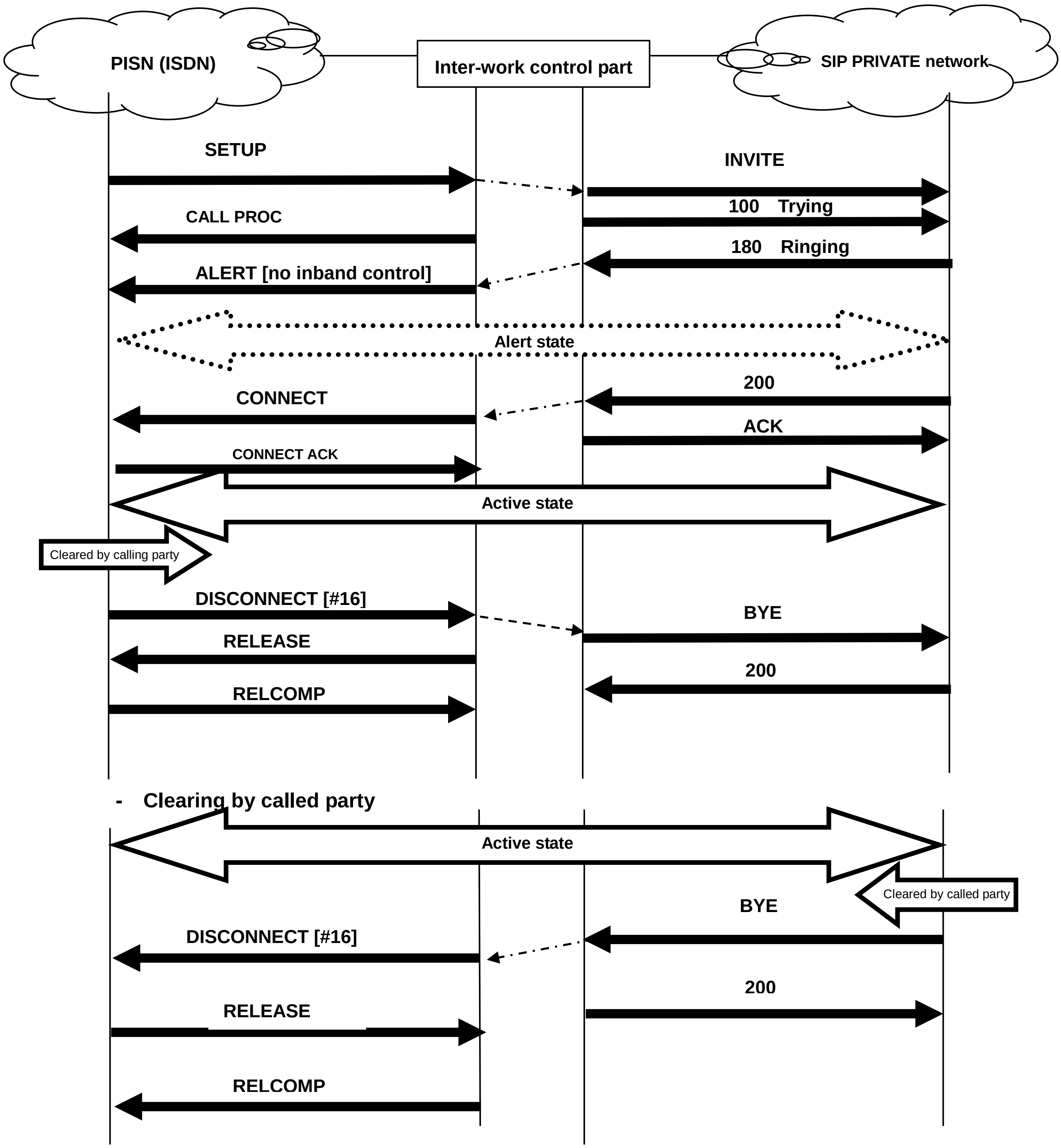


Figure 3-9/JJ-22.02 Normal sequence (transmission from PISN to SIP network)

3.4.3. Normal sequence (transmission from SIP private network to PSIN), early media control performed

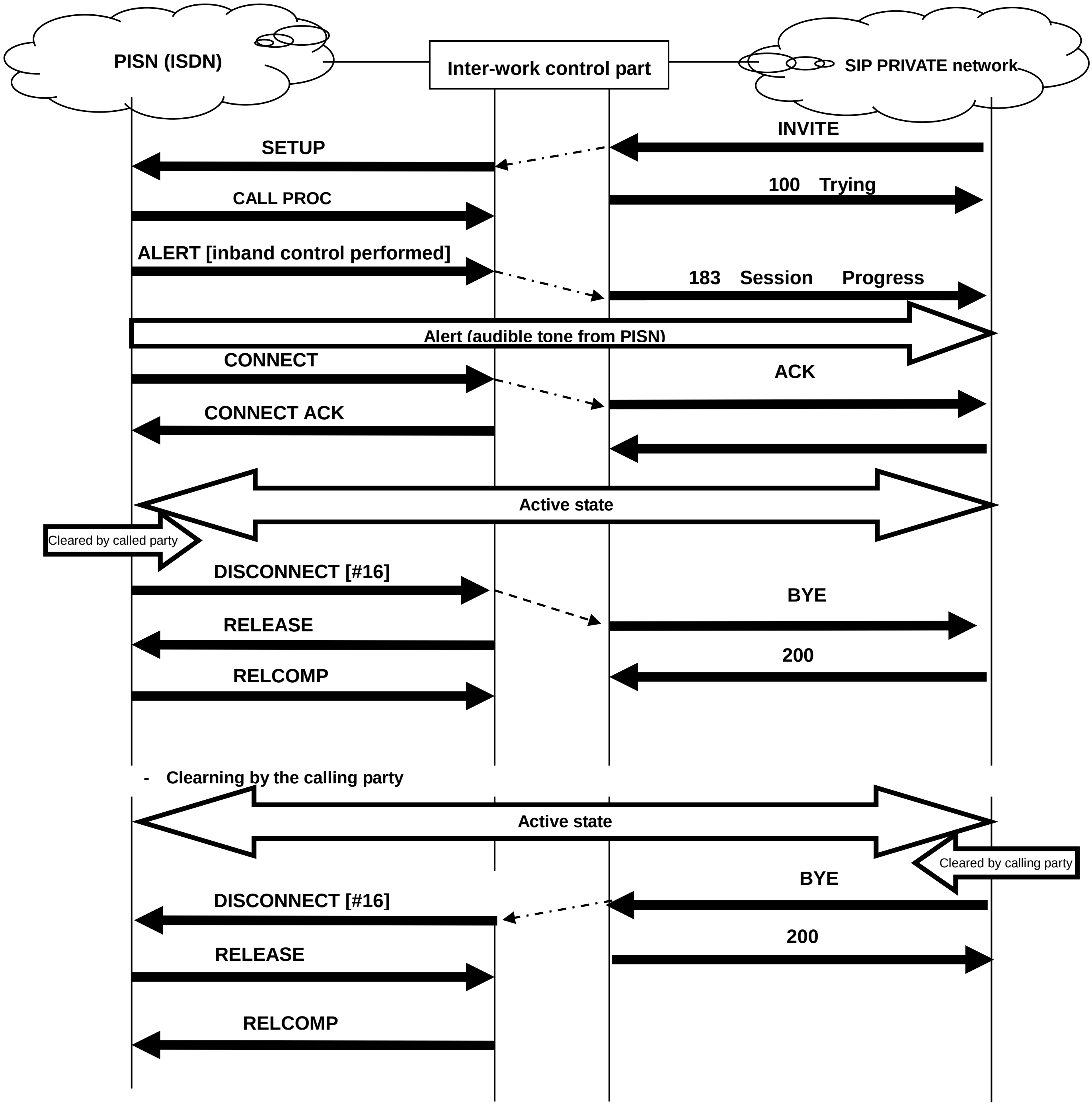


Figure 3-10/JJ-22.02 Normal sequence (transmission from SIP private network to PSIN)

3.4.4. Normal sequence (transmission from PISN to SIP network), early media control performed

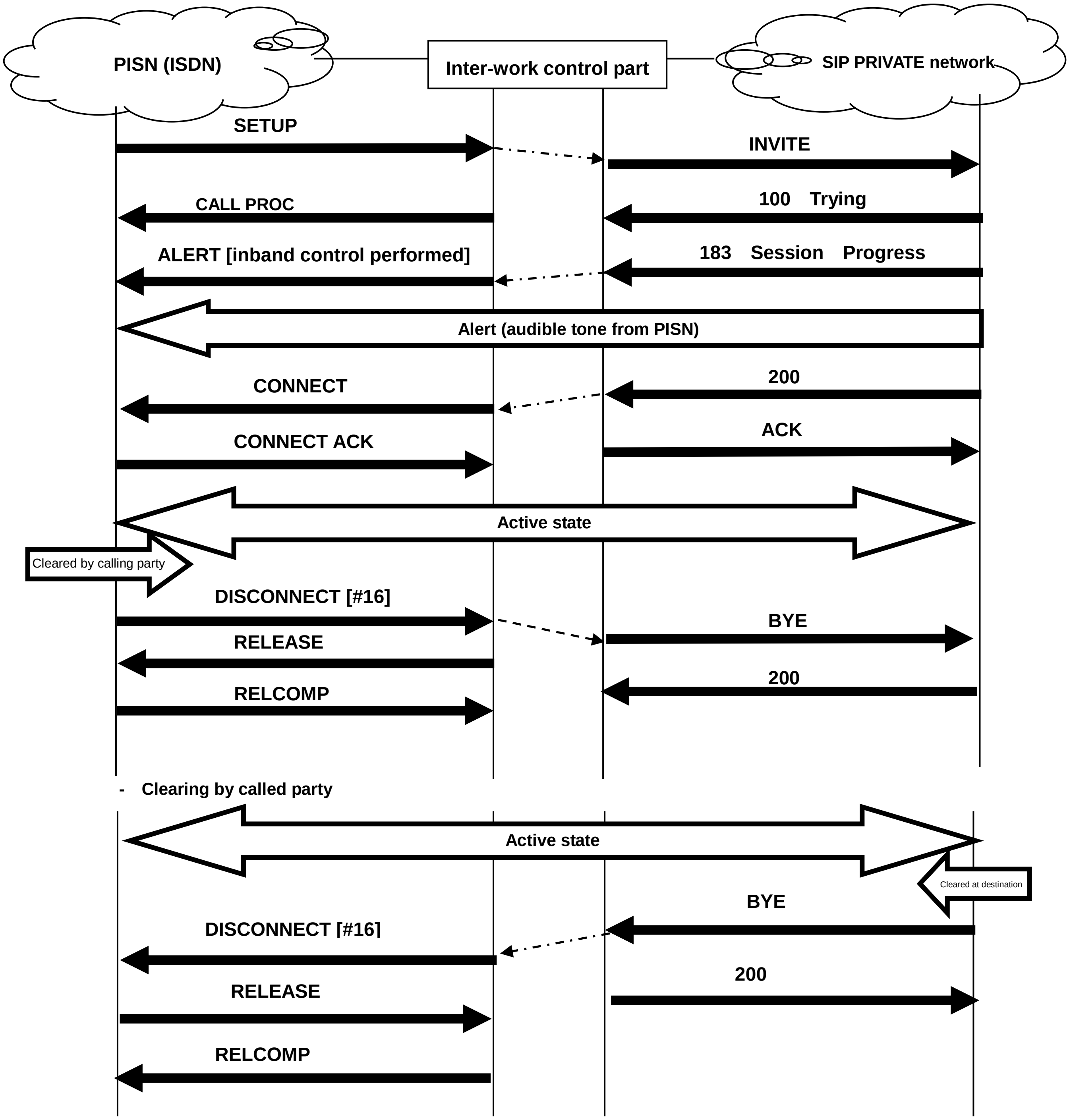


Figure 3-11/JJ-22.02 Normal sequence (transmission from PISN to SIP network)

3.4.5. Quasi-normal sequence (transmission from SIP network to PISN: abort)

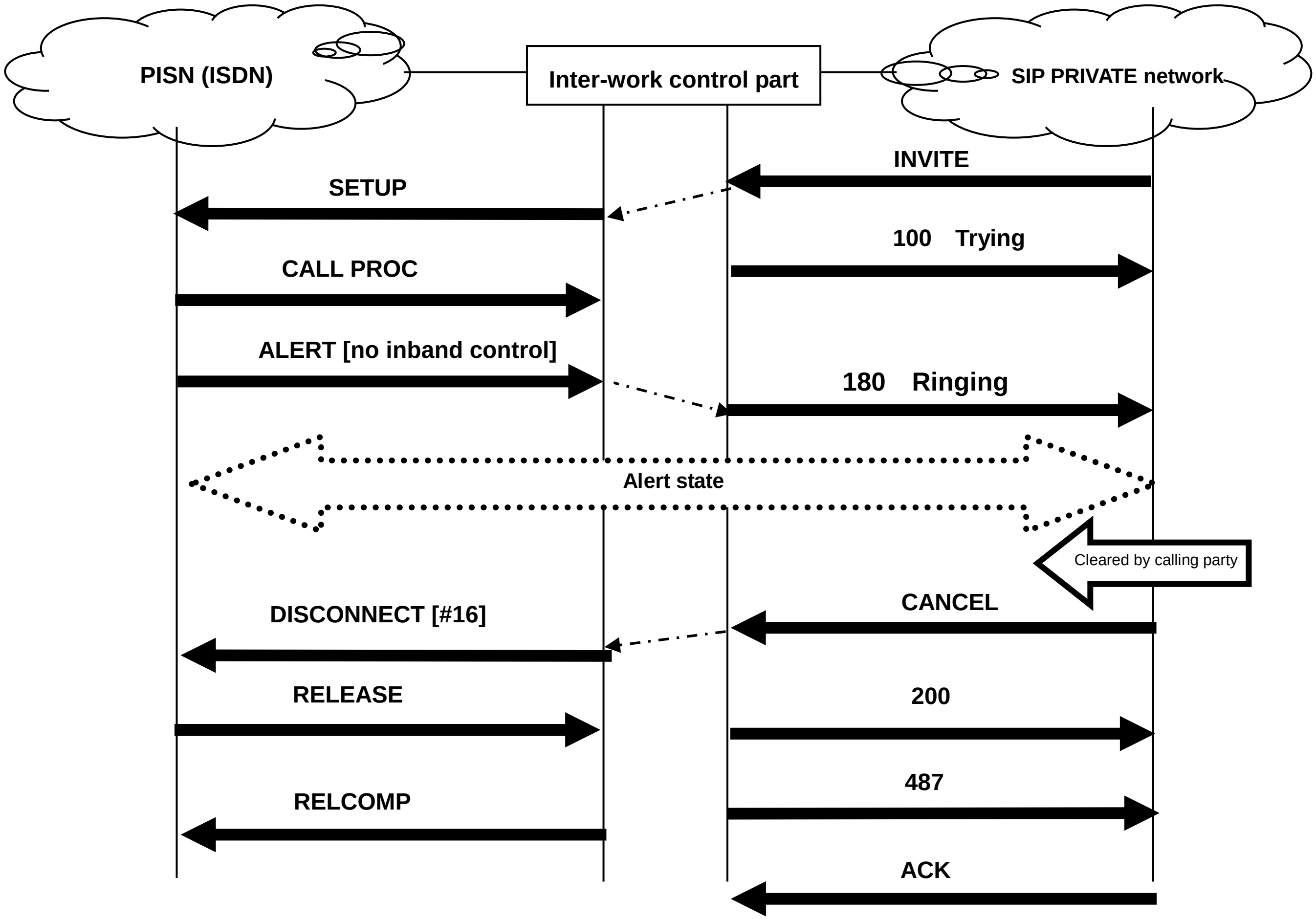


Figure 3-12/JJ-22.02 Quasi-normal sequence (transmission from SIP network to PISN: abort)

3.4.6. Quasi-normal sequence (transmission from PISN to SIP network: abort)

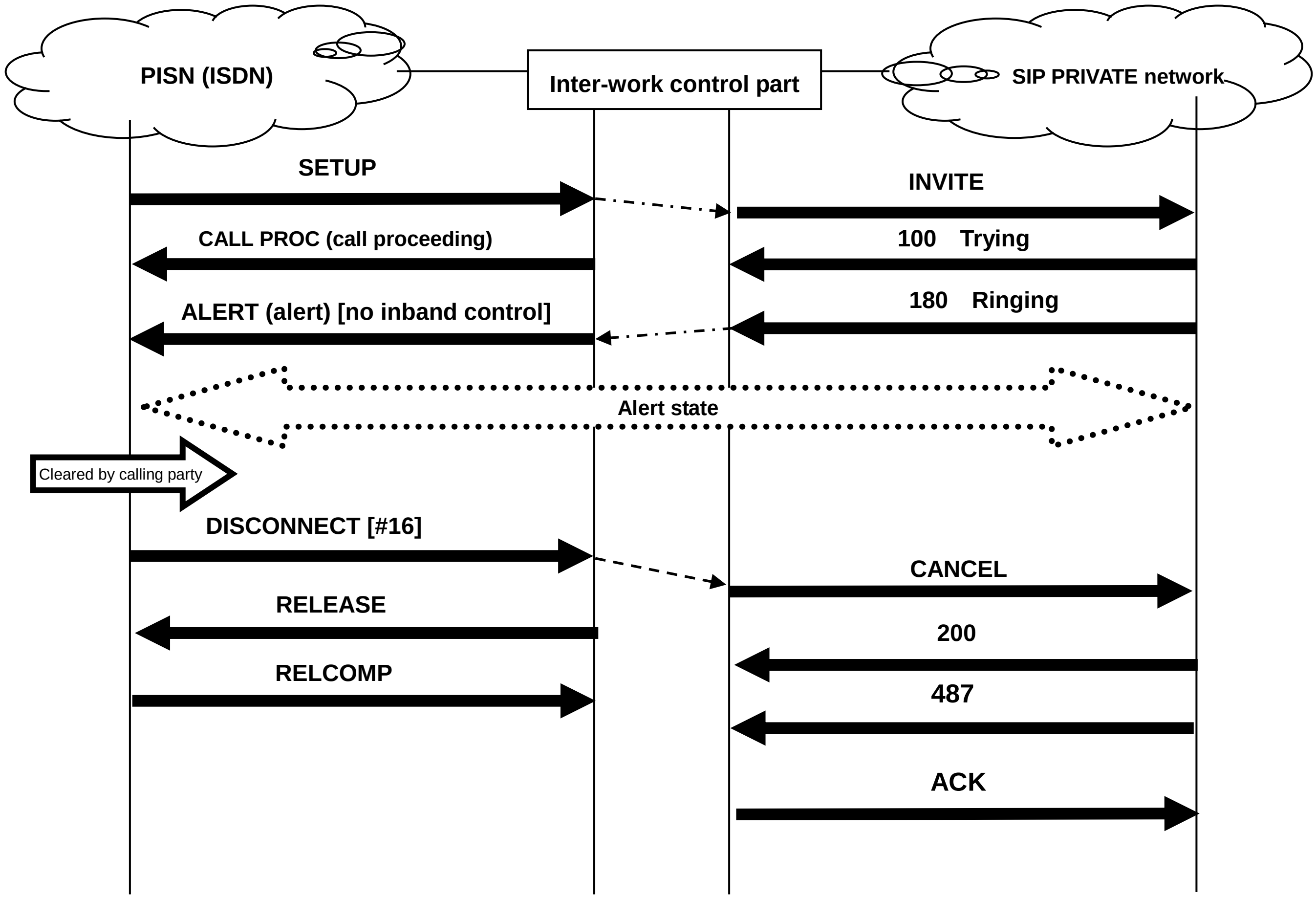


Figure 3-13/JJ-22.02 Quasi-normal sequence (transmission from PISN to SIP network: abort)

3.4.7. Quasi-normal sequence

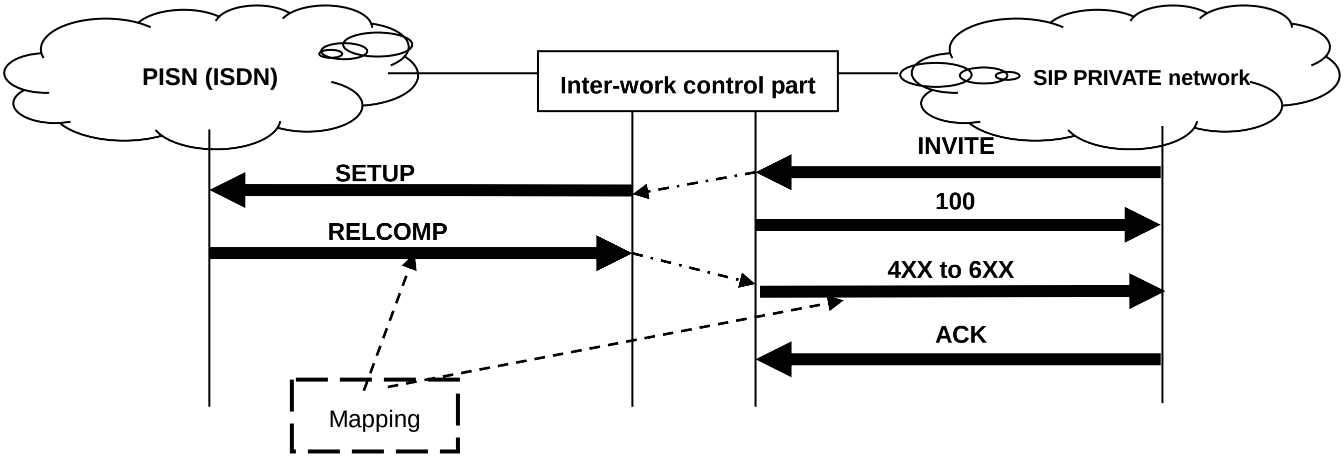


Figure 3-14/JJ-22.02 Quasi-normal sequence error response sequence example 1

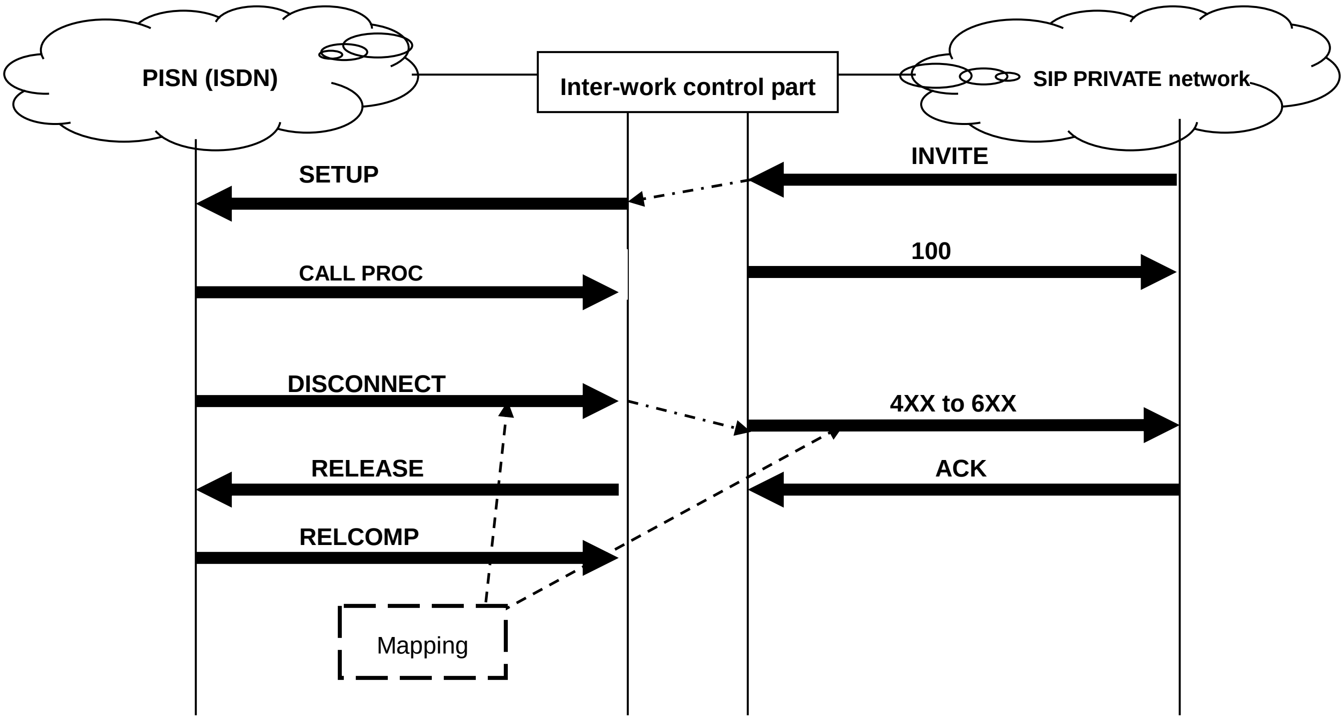


Figure 3-15/JJ-22.02 Quasi-normal sequence error response sequence example 2

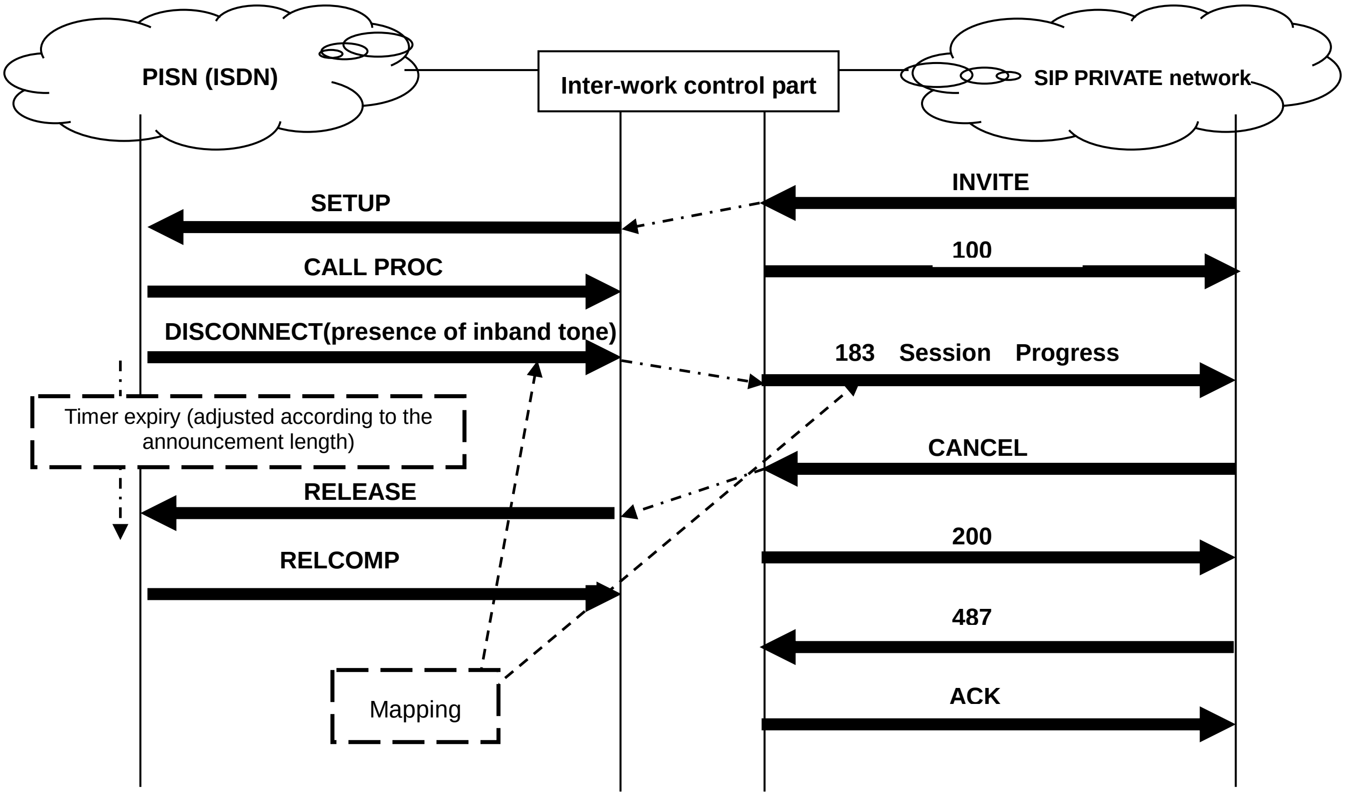


Figure 3-16/JJ-22.02 Quasi-normal sequence error response sequence example 3

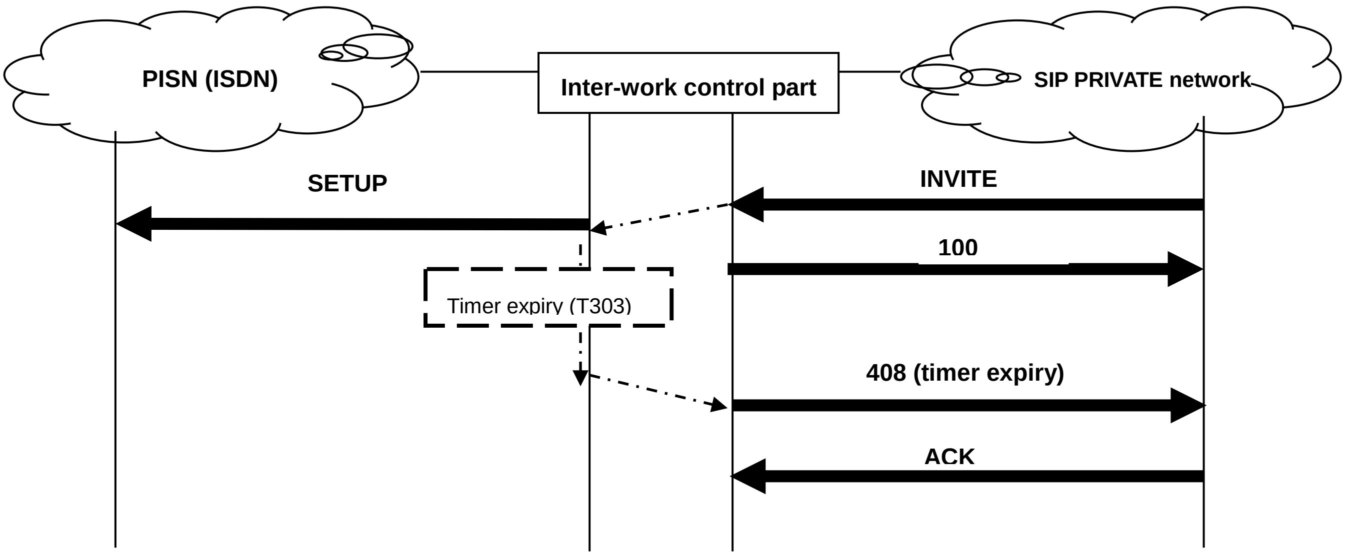


Figure 3-17/JJ-22.02 Quasi-normal sequence error response sequence example 4

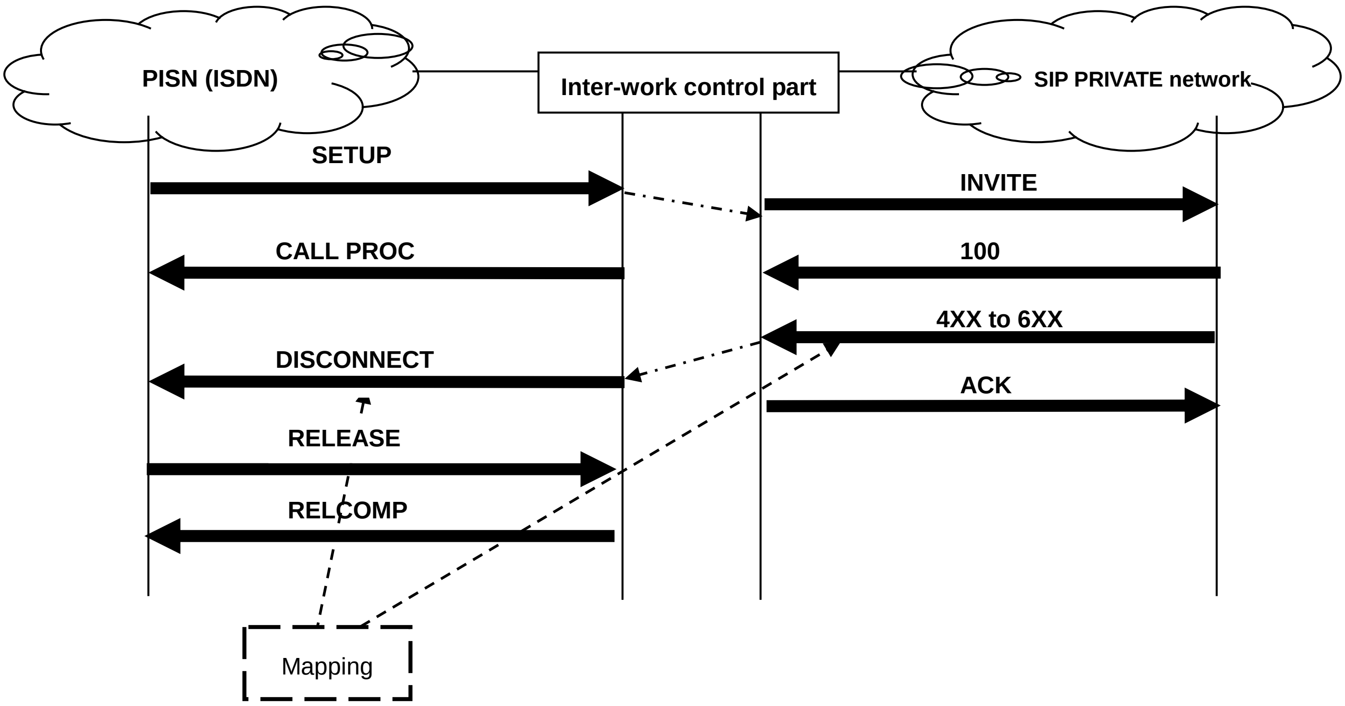


Figure 3-18/JJ-22.02 Quasi-normal sequence error response sequence example 5

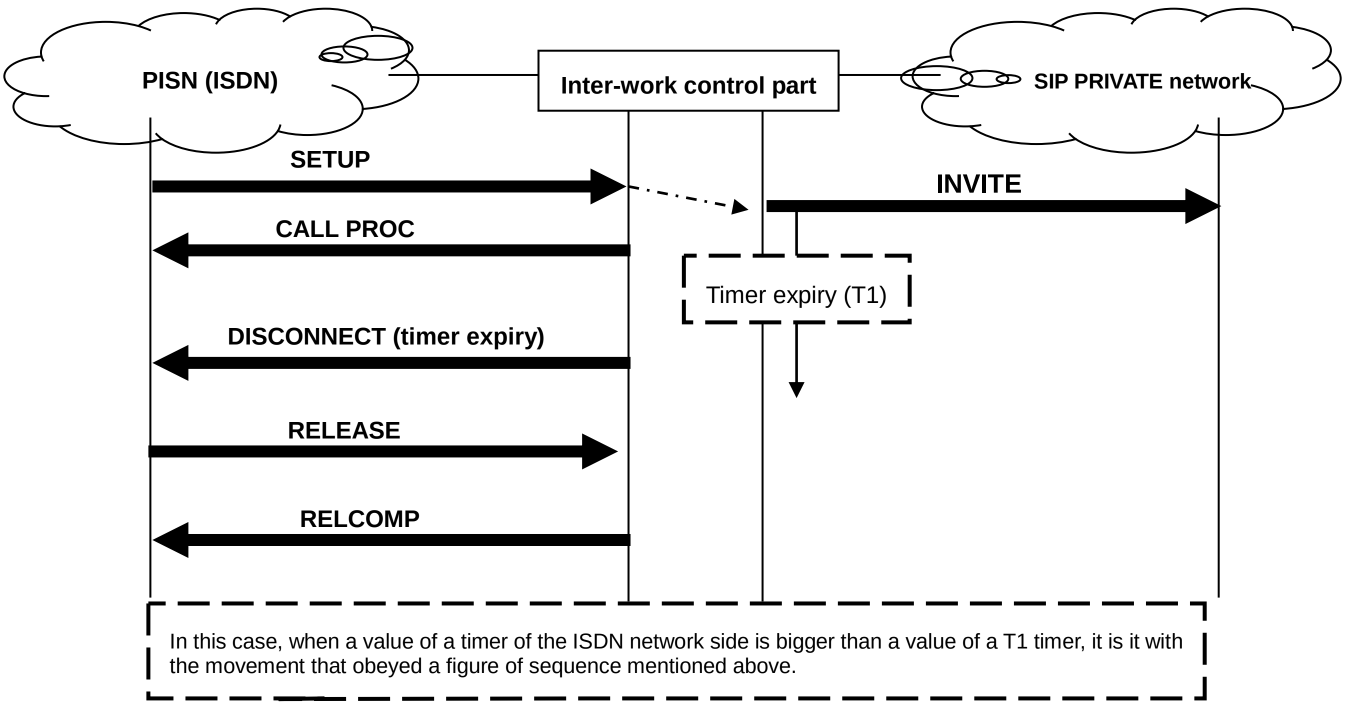


Figure 3-19/JJ-22.02 Quasi-normal sequence error response sequence example 6

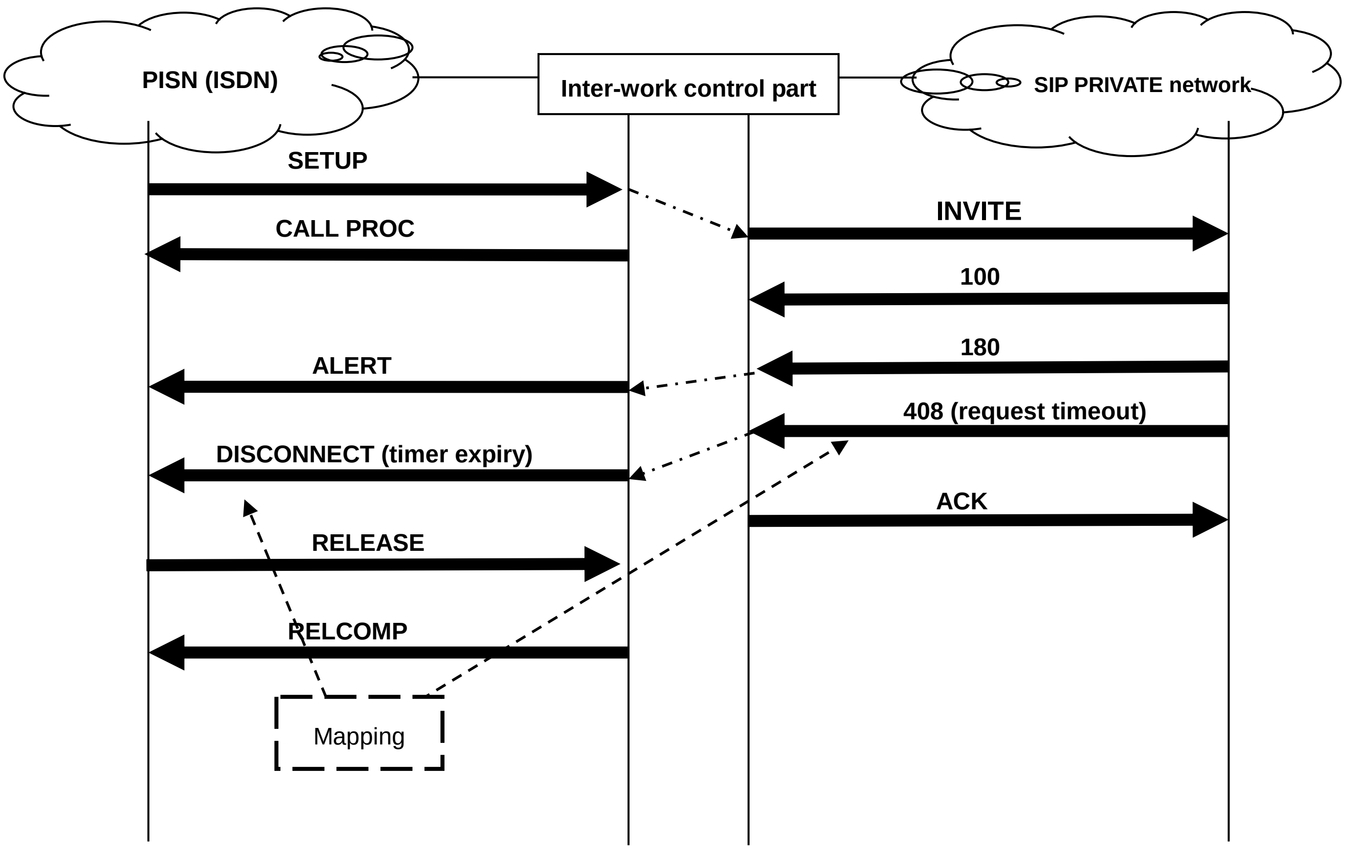


Figure 3-20/JJ-22.02 Quasi-normal sequence error response sequence example 7

4. Timer-Related Information

If protocols with different signalling systems are used, operation is often affected by the protocol timers among the protocols and, therefore, timer provisions applicable when the protocols are interconnected are required to maintain affinity. Thus, this section describes the desirable methods to set timer values.

4.1. Incoming (JS-11572), outgoing (SIP)

Conditions applied if the incoming side is specified as the side that receives (terminates) JS-11572 messages and the outgoing side is specified as the side that sends (originates) SIP messages

Table 4-1/JJ-22.02 Timer (incoming (JS-11572), outgoing (SIP))

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	JS-11572 (ISDN)			SIP			
1	T301	-	Timer in implement matter alert state	Timer-X	O	Timer in alert state	The timer should be entrusted with each network and, therefore, the inter-work part should not set this timer.

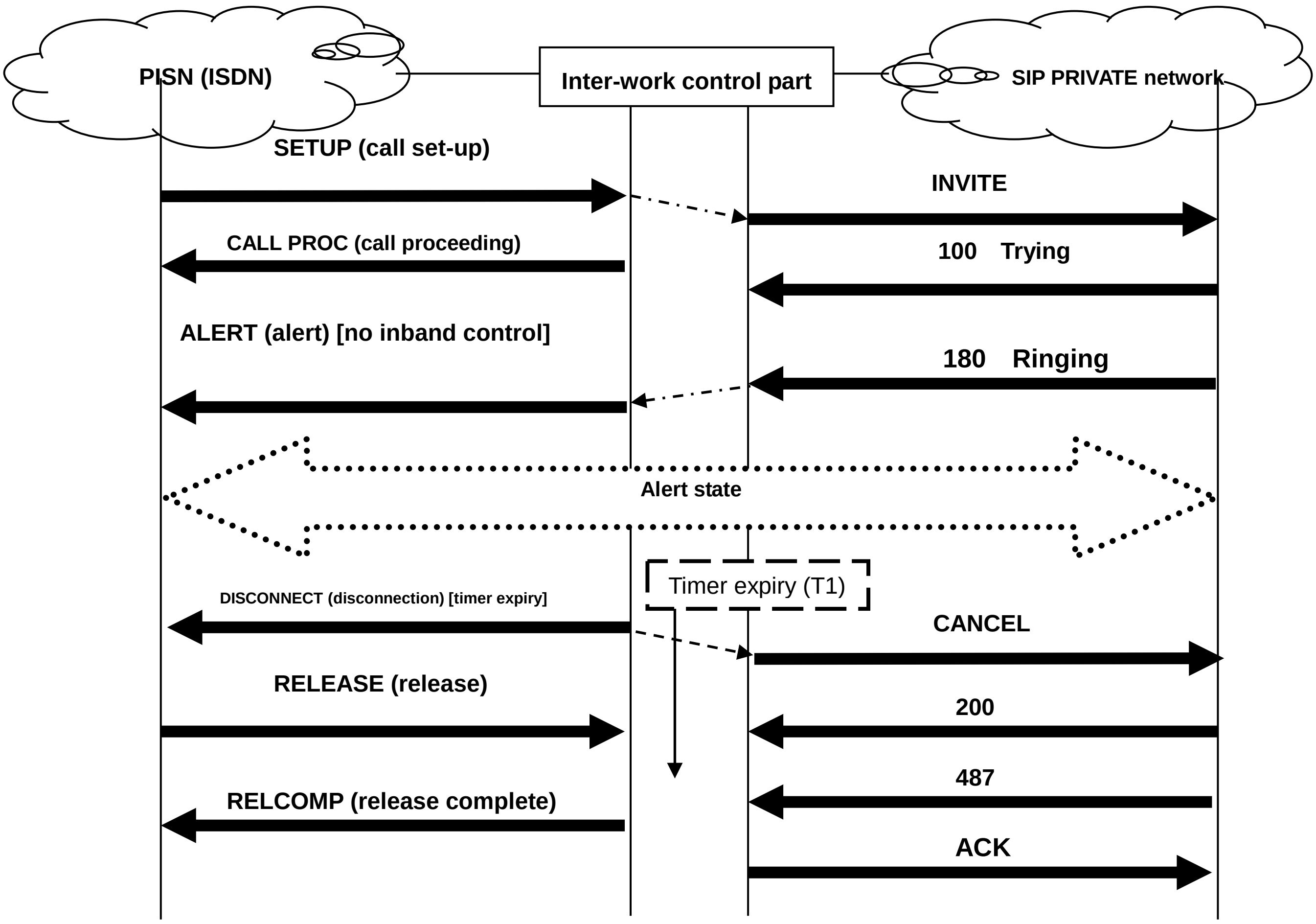


Figure 4-1/JJ-22.02 Timer (incoming (JS-11572), outgoing (SIP))

Table 4-2/JJ-22.02 Timer (incoming (SIP), outgoing (JS-11572))

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
2	---	-	None	T302	M	14s to 16s Overlap receiving	Overlap receiving on SIP is currently not considered.

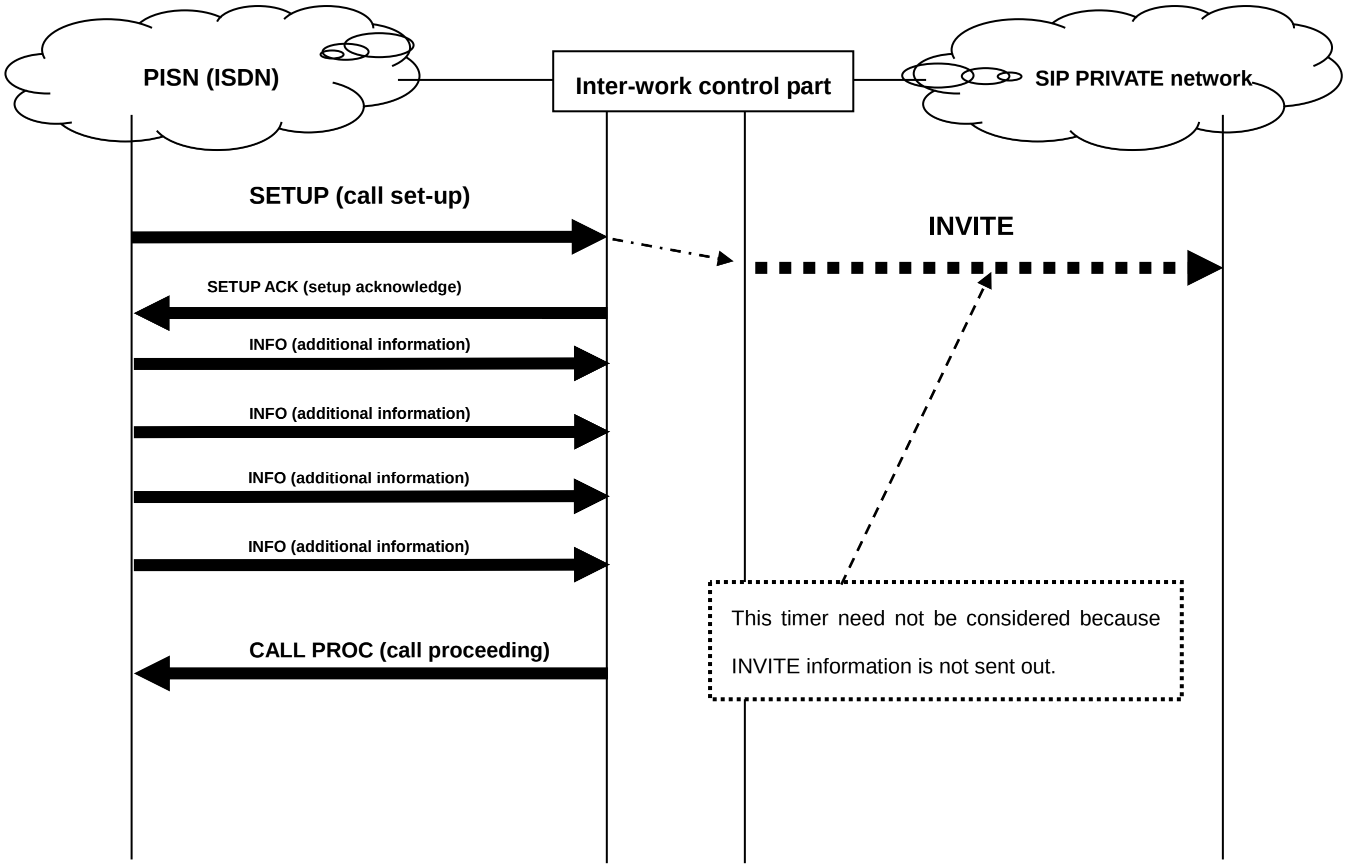


Figure 4-2/JJ-22.02 Timer (incoming (SIP), outgoing (JS-11572))

Table 4-3/JJ-22.02 Timer expiry (incoming(SIP), outgoing (JS-11572))

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
3	T303	-	4s to 6s (first time) 4s to 6s (second time) option	Timer-F	M	INVITE retransmission timer	This timer must be set to at least four seconds. Considering the T310 timer on the PISN, the maximum value should not exceed the value of T310. The inter-work part may be detected as a failure because of the timer expiry at the calling party.

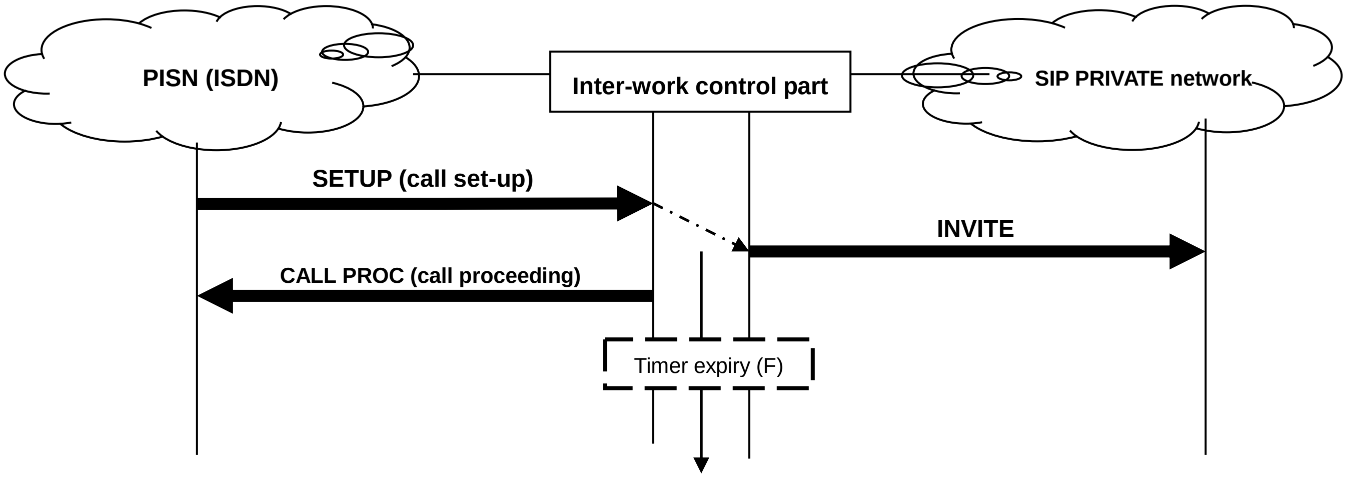


Figure 4-3/JJ-22.02 Timer expiry (incoming (SIP), outgoing (JS-11572))

Table 4-4/JJ-22.02 Timer (interaction between JS-11572 (ISDN) and SIP)

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
4	---	-	None	T304	-	20s minimum Overlap sending	Not considered because only enbloc call setup on the SIP side is supported.
5	T2	M	Non-INVITE-time request timer	T305	M	4s to 30s Clear request	Need not be considered in the inter-work part.
6	T2	M	Non-INVITE-time request timer	T308	M	4s to 6s (first time) 4s to 6s (second time) Release request	Need not be considered in the inter-work part.
7	---	-	None	T309	M	90s Any state	The SIP side is to clear the call at T309 timer expiry.
8	---	-	None	T310	-	30s or greater Outgoing call proceeding	Not applicable
9	T2	-	Non-INVITE-time request timer	T313	O	4s to 6s Response request	Whether to clear the call is regarded as an implement matter.
10	---	-	None	T316	M	120s 120s (final) Restart request	Not applicable
11	Session timer	O	Session timer	T322	M	4s to 6s Not idle	The call is to be cleared when the inter-work part sends a status inquiry message and the timer expires. Also, at the expiry of the session timer, the PISN side is prompted to release the call.

4.2. Incoming (SIP), outgoing (JS-11572)

The incoming side is regarded as the side that terminates SIP messages, while the outgoing side is regarded as the side that sends (originates) JS-11572 messages.

Table 4-5/JJ-22.02 Implement matter timer (incoming (SIP), outgoing (JS-11572))

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
1	Timer-X	-	Timer in alert state	T301	O	Implement matter Timer in alert state	The timer should be entrusted with each network and, therefore, the inter-work part should not set this timer.

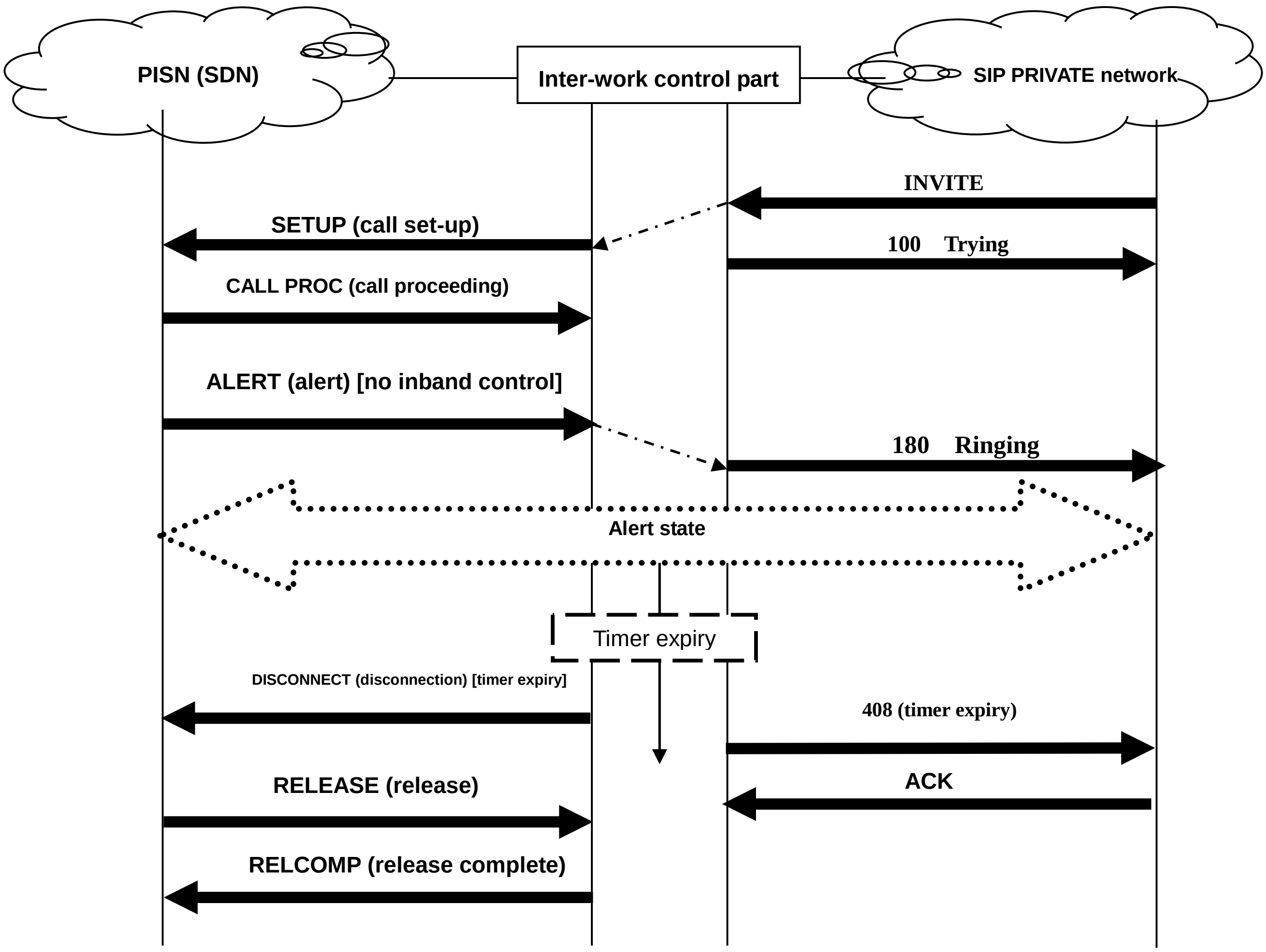


Figure 4-4/JJ-22.02 Implement matter timer (incoming (SIP), outgoing (JS-11572))

Table 4-6/JJ-22.02　Timer overlap receiving (incoming (SIP), outgoing (JS-11572))

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
2	———	-	None	T302	M	14s to 16s Overlap receiving	Overlap receiving on the SIP side is currently not considered.

Table 4-7/JJ-22.02　Timer expiry (incoming (SIP), outgoing (JS-11572)T303)

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
3	Timer-F	M	INVITEretransmission timer	T303	-	4s to 6s (first time) 4s to 6s (second time) option	The timer in the inter-work part must be a timer of least ten seconds.

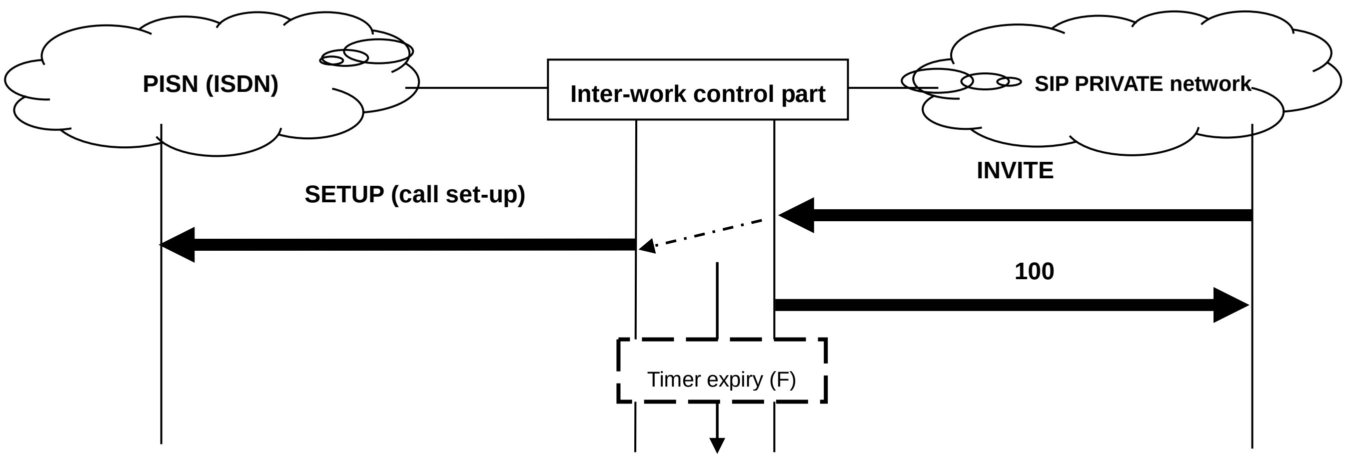


Figure 4-5/JJ-22.02　Timer expiry (incoming (SIP), outgoing (JS-11572) T303)

Table 4-8/JJ-22.02 Interaction between JS-11572 (ISDN) and SIP

Item No.	Interaction between JS-11572 (ISDN) and SIP						Desirable setting method
	Incoming (termination)			Outgoing (origination)			
	SIP			JS-11572 (ISDN)			
4	---	-	None	T304	-	20s minimum Overlap sending	Not considered because only enbloc call setup on the SIP side is supported.
5	T2	M	Non-INVITE-time request timer	T304	M	4s to 30s Clear request	Need not be considered in the inter-work part.
6	T2	M	Non-INVITE-time request timer	T308	M	4s to 6s (first time) 4s to 6s (second time) Release request	Need not be considered in the inter-work part.
7	---	-	None	T309	M	90s Any state	The SIP side is to clear the call at T309 timer expiry.
8	---	-	None	T310	-	30s or greater Outgoing call proceeding	Not applicable
9	T2	-	Non-INVITE-time request timer	T313	O	4s to 6s Response request	Whether to clear the call is regarded as an implement matter.
10	---	-	None	T316	M	120s 120s (final) Restart request	Not applicable
11	Session timer	O	Session timer	T322	M	4s to 6s Not idle	The call is to be cleared when the inter-work part sends a status inquiry message and the timer expires. Also, at the expiry of the session timer, the PISN side is prompted to release the call.

5. Local Error Response Values of the Inter-work Part

This chapter describes the definitions of the local error response values of the inter-work part. The aim of these definitions is to enable the SIP private network to clearly determine at which reference point an error has been generated, by clearly distinguishing local error response values from global error response codes, thereby assisting in isolating failures and so on.

5.1. Signal sequence example

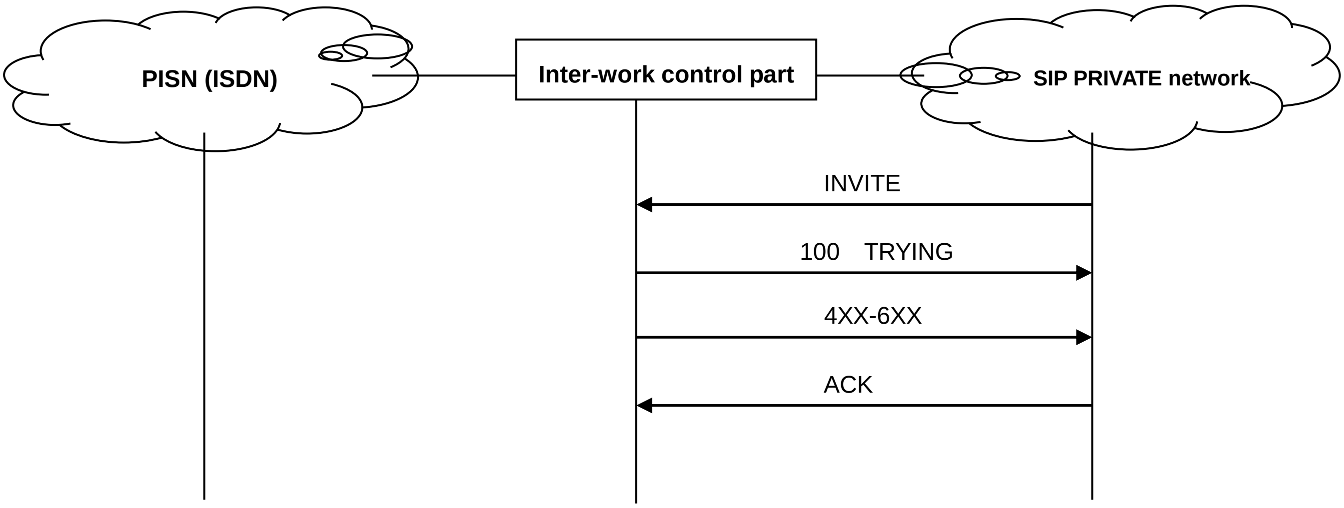


Figure 5-1/JJ-22.02 Error response in the inter-work part, example 1

5.2. Internal error response values in the inter-work part

Table 4-8/JJ-22.02 Mapping from GW device to SIP

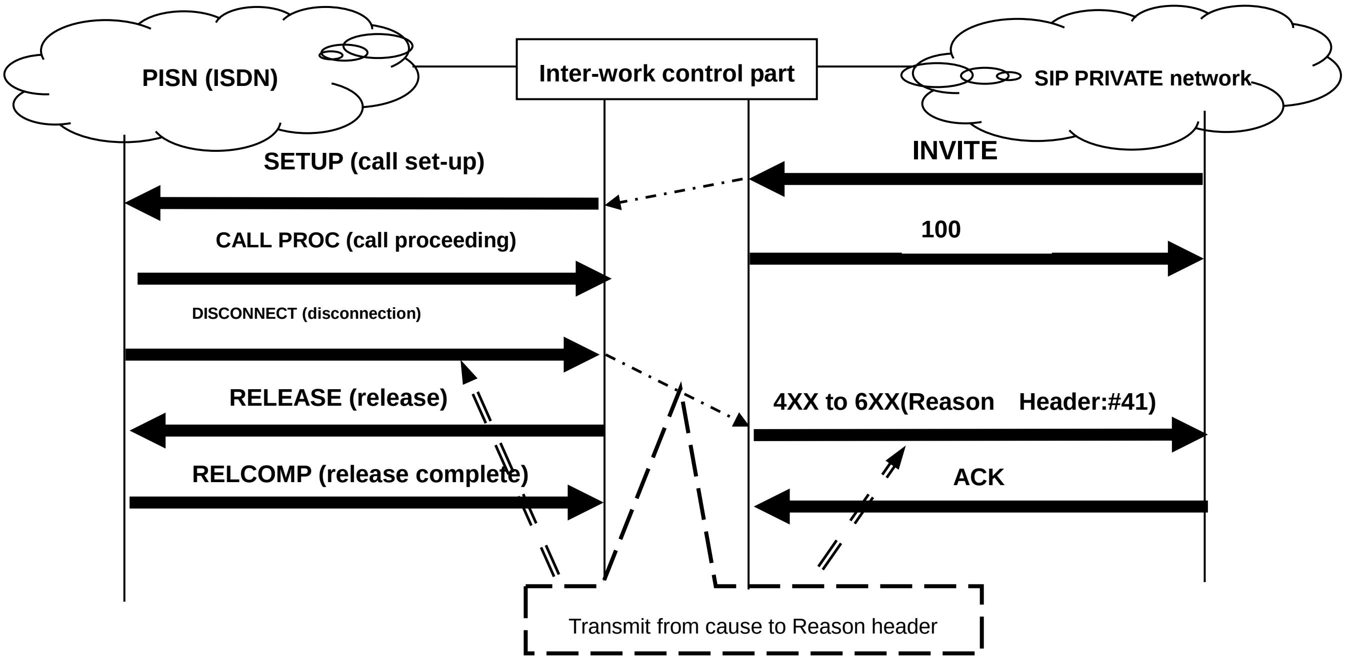
Item No.	Mapping from GW device to SIP		Remarks
	GW	SIP	
1	All channels busy	503	
2	No resources	503	
3	GW internal congestion	503	

Other error responses shall be regarded as implement matters.

6. Support of the Reason Header (RFC3326)

Because response values cannot be passed between ISDN and SIP, the Reason header shall be used to cause response value with Q850 to transmit.

If Reason header information is carried on response information and received, the Reason header information should be checked first. Whether to carry the Reason header shall be entrusted with the implementation of the GW/server at the terminating point.



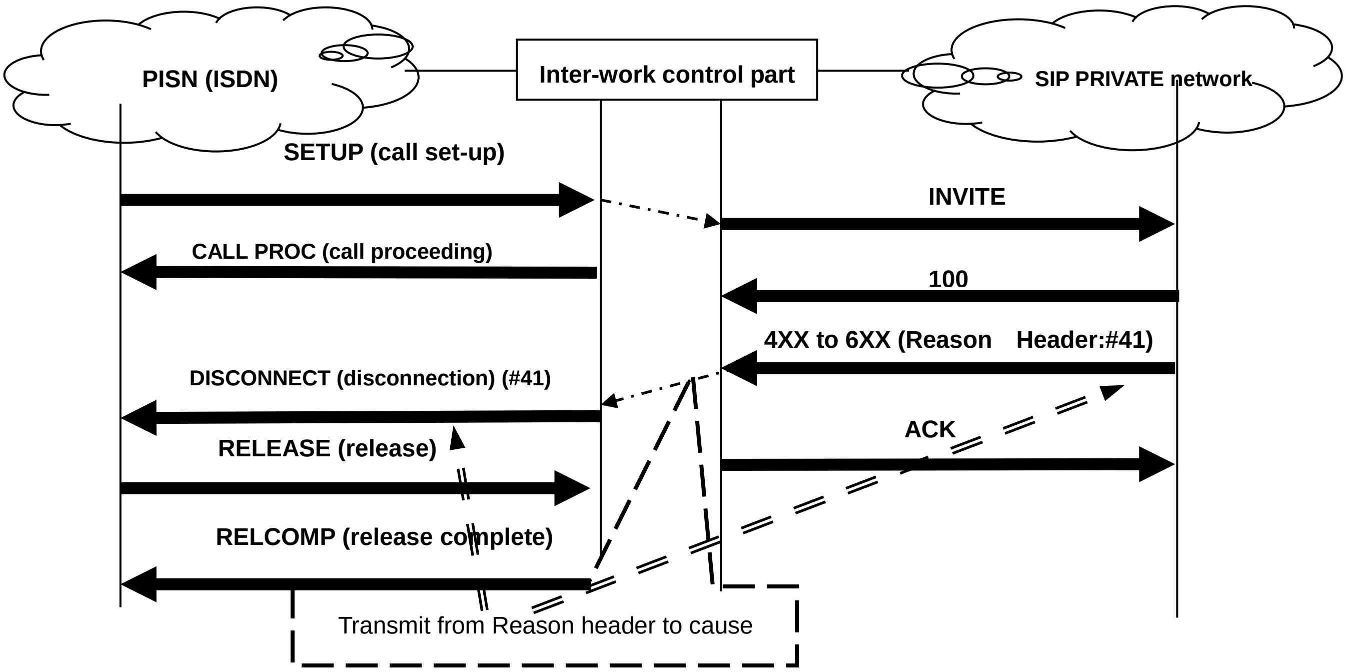


Figure 6-1/JJ-22.02 Reason Header

Coding information for the Reason header are given below.

```
Reason              =   "Reason" HCOLON reason-value *(COMMA reason-value)
reason-value        =   protocol *(SEMI reason-params)
protocol            =   "SIP" / "Q.850" / token
reason-params       =   protocol-cause / reason-text
                        / reason-extension
protocol-cause      =   "cause" EQUAL cause
cause               =   1*DIGIT
reason-extension    =   generic-param
```

SIP: The cause parameter contains a SIP status code.

Q.850: The cause parameter contains an ITU-T Q.850 cause value

in decimal representation.

Examples are:

Reason: SIP ;cause=200 ;text="Call completed elsewhere"

Reason: Q.850 ;cause=16 ;text="Terminated"

Reason: SIP ;cause=600 ;text="Busy Everywhere"

Reason: SIP ;cause=580 ;text="Precondition Fai

7. Appendix.

Those cases can be considered in which, in inter-working with an existing PBX, the PBX provides non-ISDN lines only. In those cases, as viewed from the SIP private network, the states of the terminals managed by the PBX may not be conveyed to the SIP network, making it impossible to determine whether the terminals are busy or idle.

Also, as for non-ISDN lines, the inter-work part must have the capability for early media control; otherwise, the terminal states will not be conveyed to the end point.

Thus, it is desirable that the inter-work device that responds to individual user needs be provided with a protocol supporting early media.

Inter-work Specifications between Private SIP Network and private ISDN Network

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