TTC標準
Standard

JJ-22.06

Technical Specification Call Transfer Supplementary Services Information Interface between Private SIP Networks

Version 1.2

June 9, 2016

THE TELECOMMUNICATION TECHNOLOGY COMMITTEE

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1. Introduction

The Private Network Interface Sub-Working Group of the Private Network Special Committee has been standardizing IP protocols intended for inter-PBX (Private Branch eXchange) private networks (circuit-switched networks) and Qsig (Signaling information flows at the Qreference point). Now, taking recent trends of the market and international recommendations into consideration, it is necessary to study implementation of VoIP (Voice over Internet Protocol) technology based on SIP (Session Initiation Protocol) in operating agencies. This standard specification is stated, focusing on how the new technologies mentioned above are lately handled and how carriers are handling them.

This standard, which is referring to JS-22535 (Qsig tunneling), is a material for standardization of making Qsig-based inter-office services also available in the SIP (Session Initiation Protocol) network.
This standard is a material specially dedicated for "call transfer supplementary services".

2. Revision History

Version	Date of establishment	Description
First Version	May 27, 2009	Established.
Version 1.1	December 10,2009	Revision.
Version 1.2	June 9,2016	Revision (Correction of Figure 2.4.1.1)

3. Miscellaneous

(1) Recommendations, standards, etc., referenced

JS-13869 : Private Integrated Services Network (Call Transfer supplementary service)
-Specifications for inter-PBX signaling protocol -

JS-11572 : Private Integrated Services Network (circuit-mode bearer services)
- Layer 3 Specifications for inter-PBX signaling protocol -

JS-11582 : Private Integrated Services Network (Generic Functional Protocol for the support of supplementary services)
- Specifications for inter-PBX signaling protocol -

JS-22535 : Technical Specification on "QSIG" tunneling by Session Initiation Protocol (SIP) in corporate telephonic network (CN)

TTC standard : JJ-22.00 The Guideline for the Architecture of the Technical Specifications for Private SIP in TTC

TTC standard : JJ-22.01 Technical Specifications on Inter-connection Interface between Private SIP Networks

TTC standard : JJ-22.02 Inter-work Specifications between Private SIP Network and private ISDN Network

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. Outline of This Specification

This specification is a material for standardization of call transfer supplementary services in conformity with the JS-22535 (Qsig tunneling) in private SIP networks.

1.1 Purpose

This standard is intended to state definitions of inter-office services used in networks connected through an IP network (SIP) in order to plot interwork affinity and expandability of inter-office services.

1.2 Summary

This standard states the conditions for tunneling with SIP (Session Initiation Protocol), based on JS-13869 (private integrated service network (call transfer supplementary services), inter-PBX signal protocol specifications).

2. Description of the Standard

2.1 Definition of the standard

This standard stipulates "call transfer supplementary services" employing inter-office service tunneling with Session Initiation Protocol (SIP) on the private telephone communication network (CN).

SIP is an application layer protocol to start, end, and change a multimedia session. SIP is usually handled with IP transmission (RFC791, RFC2460). Telephonic calls are regarded as a type of multimedia session in which audio is exchanged. SIP is defined by RFC3261.

QSIG is a signal protocol between private integrated service network exchanges (PINX) on the private integrated service network (PISN). PISN provides circuit-switched basic services and supplementary services for users. QSIG is stipulated in the domestic standards JS-11572 (basic service call control) and JS-11582 (general-purpose function procedures for supplementary services), and call transfer supplementary services are stipulated in the domestic standard JS-13869 (private integrated service network (call transfer supplementary services), inter-PBX signal protocol specifications) and standards of respective supplementary services.

Note: QSIG is named after signaling at Q reference points. The Q reference point is the boundary between two PINXs.

A CN may consist of a PISN employing QSIG and IP network employing SIP. Calls and signals independent from calls are originated from users connecting with the PISN and terminated by a user connecting with the IP network, or vice versa. In either case, a gateway provides QSIG-SIP interworking at the boundary between PISN and IP network. The basic call interworking at the gateway is stated in ISO/IEC17343. In another case where calls and signals independent from calls are originated from users connecting with the PISN, calls are sent over an IP network employing SIP and terminated by a user connecting with another PISN (or another location in the same network).

2.2 Scope

QSIG tunneling with SIP on a public IP network is out of the scope of this standard.

This stipulation is also applicable to interworking units that function as a gateway between PISN employing QSIG and private IP network employing SIP and also provide QSIG tunneling of SIP requests/responses.

2.3 Tunneling

This document describes user-terminated calls and signals independent from calls that are originated from users connecting with a PISN employing QSIG, routed through an IP network employing SIP, and terminated by a user connecting with another PISN (or another location in the same PISN). As shown in Figure 2.3, on the boundaries between the PISNs employing QSIG and the IP network employing SIP, gateways are placed for the connections.

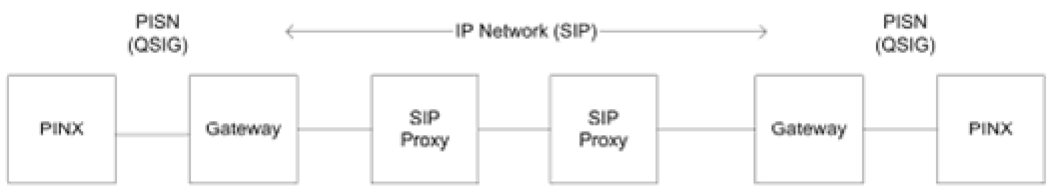


Figure 2.3 Call from QSIG to QSIG via SIP

The gateways interwork together as stated by ISO/IEC 17343. This provides basic call functions. In ISO/IEC 17343, like JS-11572, only interworking to QSIG basic calls is specified. It does not include any QSIG functions (supplementary services and additional network features) stipulated by other standards and vendor-specific specifications.

This leads to loss of functions in calls and signals independent from calls in the direction from QSIG to SIP or SIP to QSIG. Even in a case similar to that shown in Figure 2.3, loss of functions also takes place. This assumes that if the two gateways are different types only the functions common to both the gateways are available end-to-end.

As a solution to prevent loss of QSIG functions end-to-end, QSIG messages passed through the IP network are tunneled to SIP messages. Either of the two gateways starts a SIP dialog to the other gateway. By using SIP messages in the dialog, QSIG messages are tunneled. If necessary, a session is established by using SDP of RFC 3264 to transmit user information (e.g., audio) between the QSIG gateways. The two gateways function as a QSIG Transit PINX and QSIG messages are transited with little modification.

In conventional PISNs employing QSIG, associated PINXs are connected with an inter-PINX link, and it consists of one (QSIG message transmission) signal channel, and more than one user information channel for audio, modem information, or data transmission. The tunneling technique makes the IP network provide the inter-PINX link between gateways functioning as a Transit PINX. The QSIG-dedicated SIP-provided tunnel functions as a signal channel, and the media stream functions as a user information channel.

In addition to that, in case an SIP sequence failure is encountered in the QSIG-SIP interworking, it is necessary to make consideration so that no call remains on both QSIG and SIP. For example, if a timeout occurs on the SIP side, no processing is to be performed at timer-monitored locations on the end point (QSIG) side, but some processing is to be performed at other locations with an implementation-based procedure. (For example, a timeout in the middle of a sequence intended for release of a call makes it to be released and a timeout in the middle of a sequence intended for connection of a call makes it to ignore the timeout.)

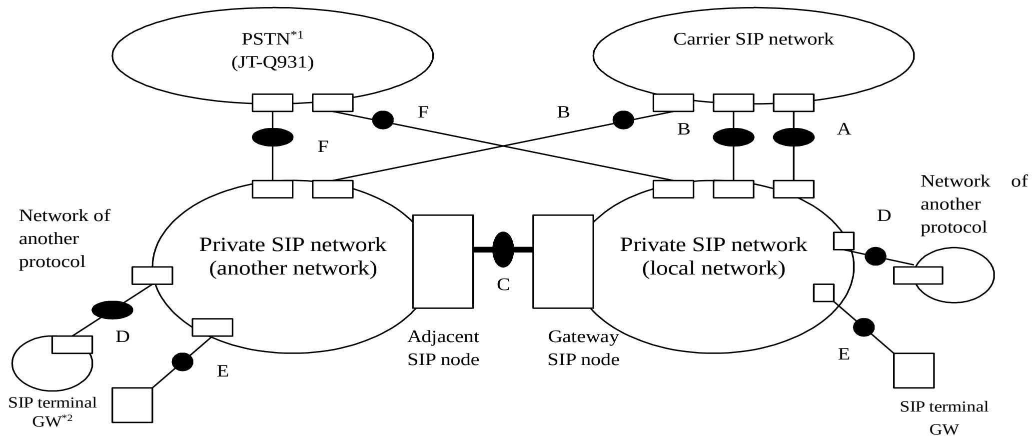
As a supplementary matter, the primary response (callproc) in tunneling is to be handled as an option.

2.4 Connection Configuration

2.4.1 Basic connection configuration

This standard describes the conditions for connection interfaces to managed private SIP networks that are applicable to Interfaces C and E specified in the private SIP network interconnection model shown in Figure 2.4.1.1.

In this standard, a private SIP network that has an interface that can observe the provisions for this interface is called a “managed private SIP network”. It is assumed in the remainder of this standard that the term private SIP network refers to a “managed private SIP network”.



*1: PSTN···Public Switched Telephone Networks
*2: GW···GateWay

Figure 2.4.1.1 Private SIP network interconnection model

3. Signal Sequences

The following abbreviations are used:

CALL PROC	CALL PROCEEDING
CONN	CONNECT
CONN ACK	CONNECT ACKNOWLEDGE
DISC	DISCONNECT
REL	RELEASE
REL COMP	RELEASE COMPLETE

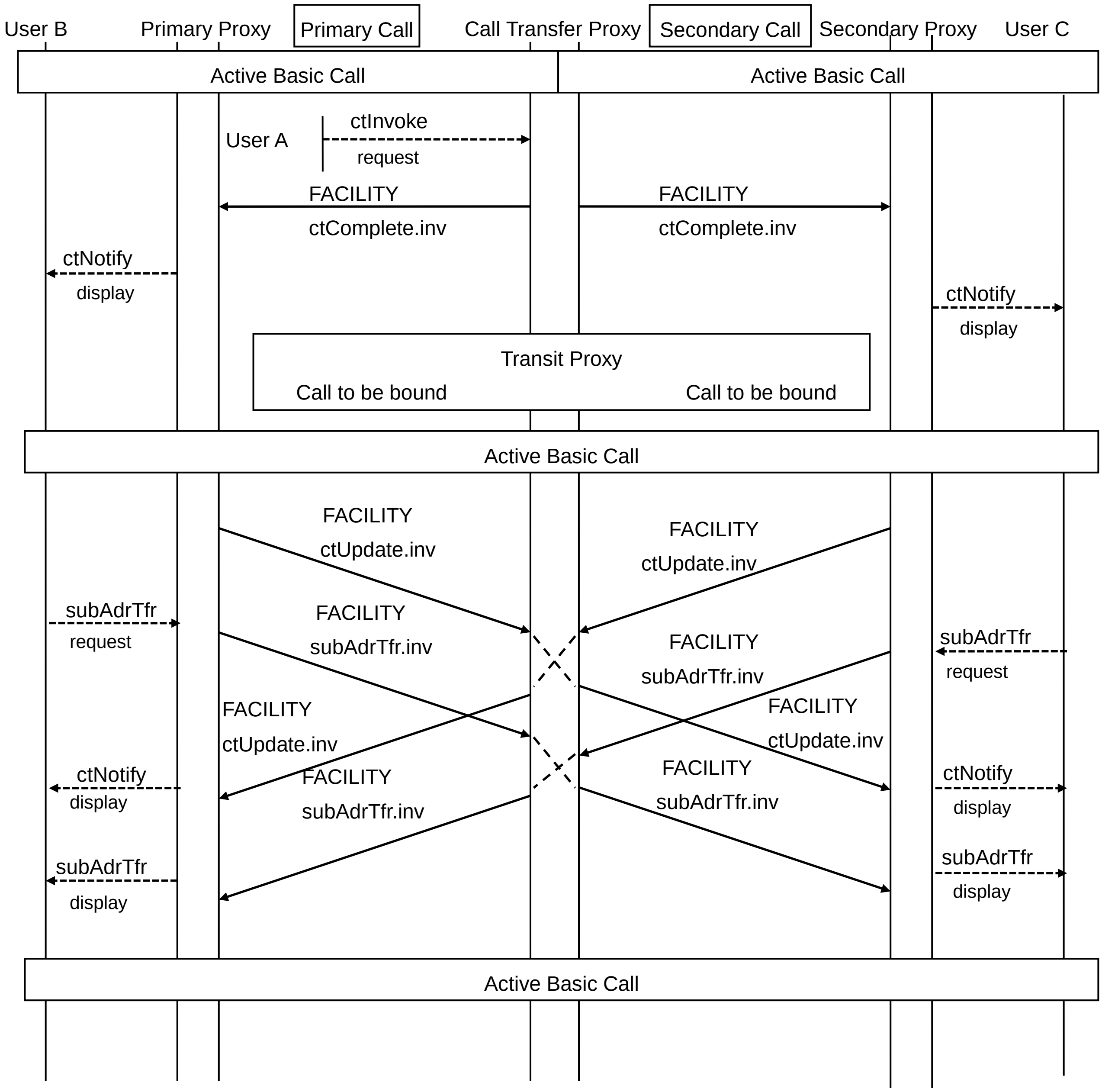
3.1 Procedures (normal sequences)

This standard states the conditions for use of interfaces (sequences) stipulated by JS-13869 with SIP messages.

(1) Example of a message sequence of normal operation of transit-type call transfer in the case where both calls are "Active"

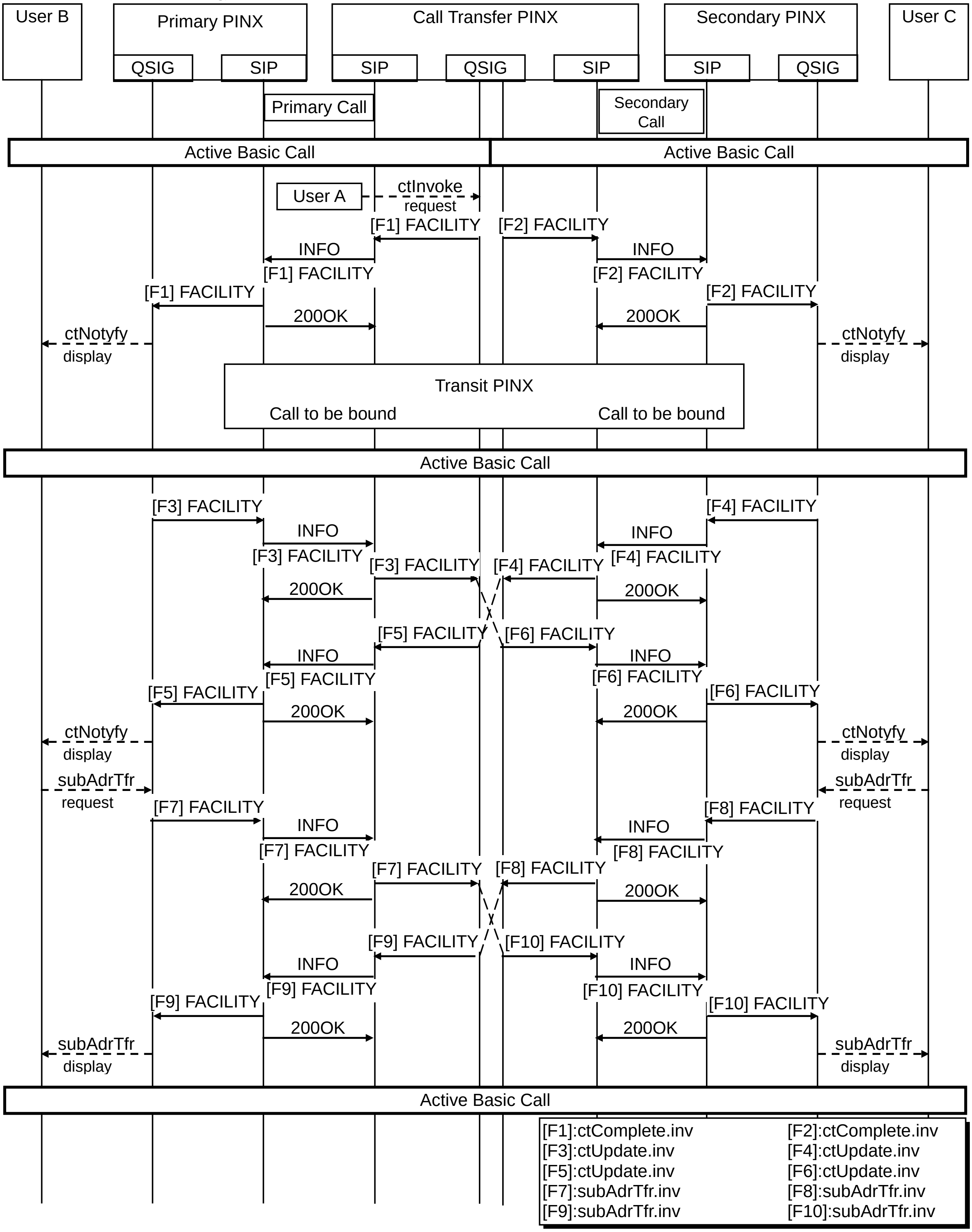
Attached Figure 3.1.1 JJ-22.06 shows an example of normal operation of transit-type call transfer in the case where both calls are "Active".

Note that the sequence example below is a quote from those defined in JS-13869.



Attached Figure 3.1.1 JJ-22.06 Message sequence of normal operation of transit-type SS-CT in the case where both calls are "Active"

(1-1) Example of a message sequence of normal operation of transit-type call transfer in the case where both calls are "Active" (in SIP tunneling)

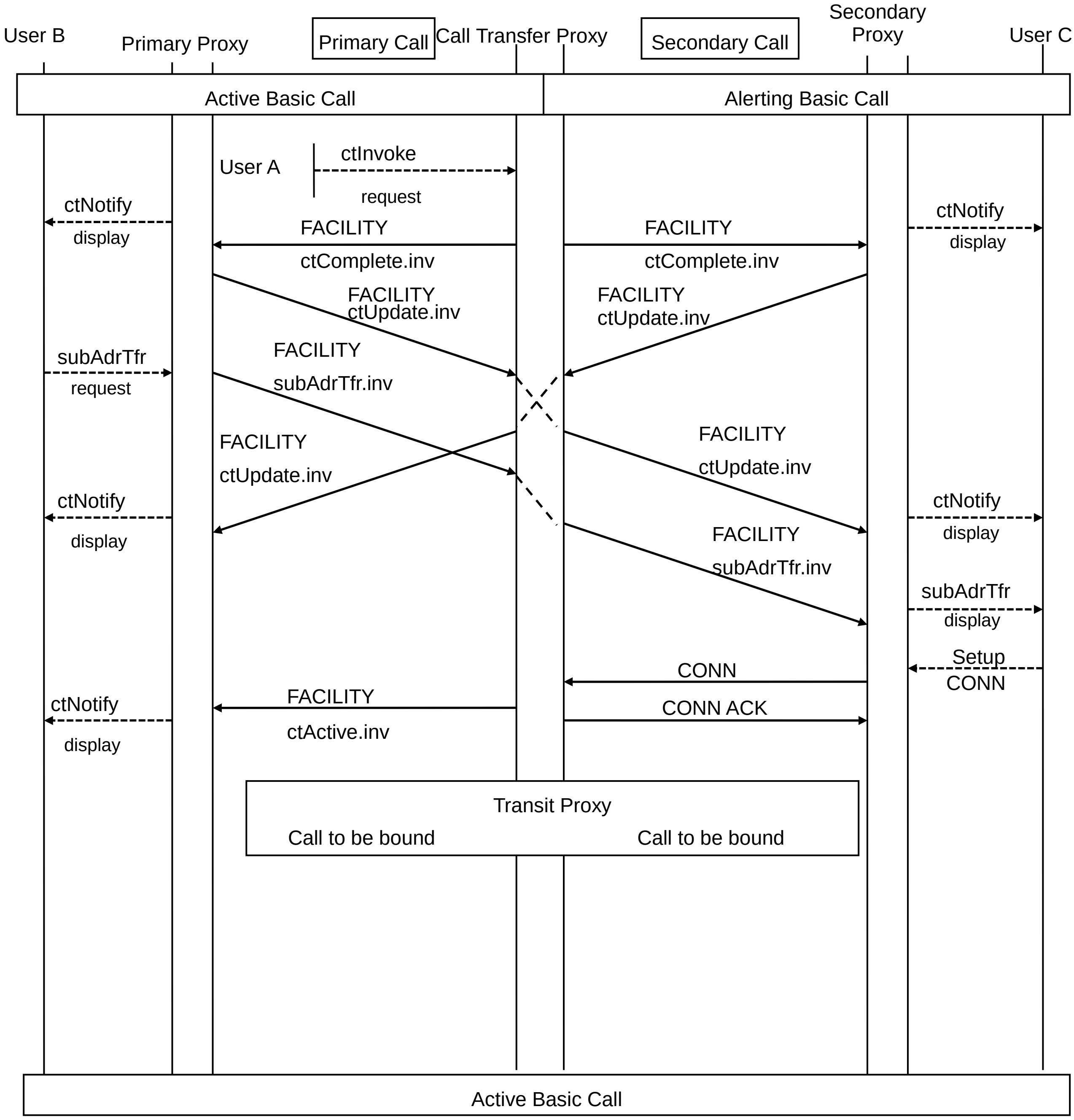


Attached Figure 3.1.2 JJ-22.06 Message sequence of normal operation of transmit-type SS-CT in the case where both calls are "Active"

(2) Example of a message sequence of transit-type call transfer in the case where one call is "Alerting"

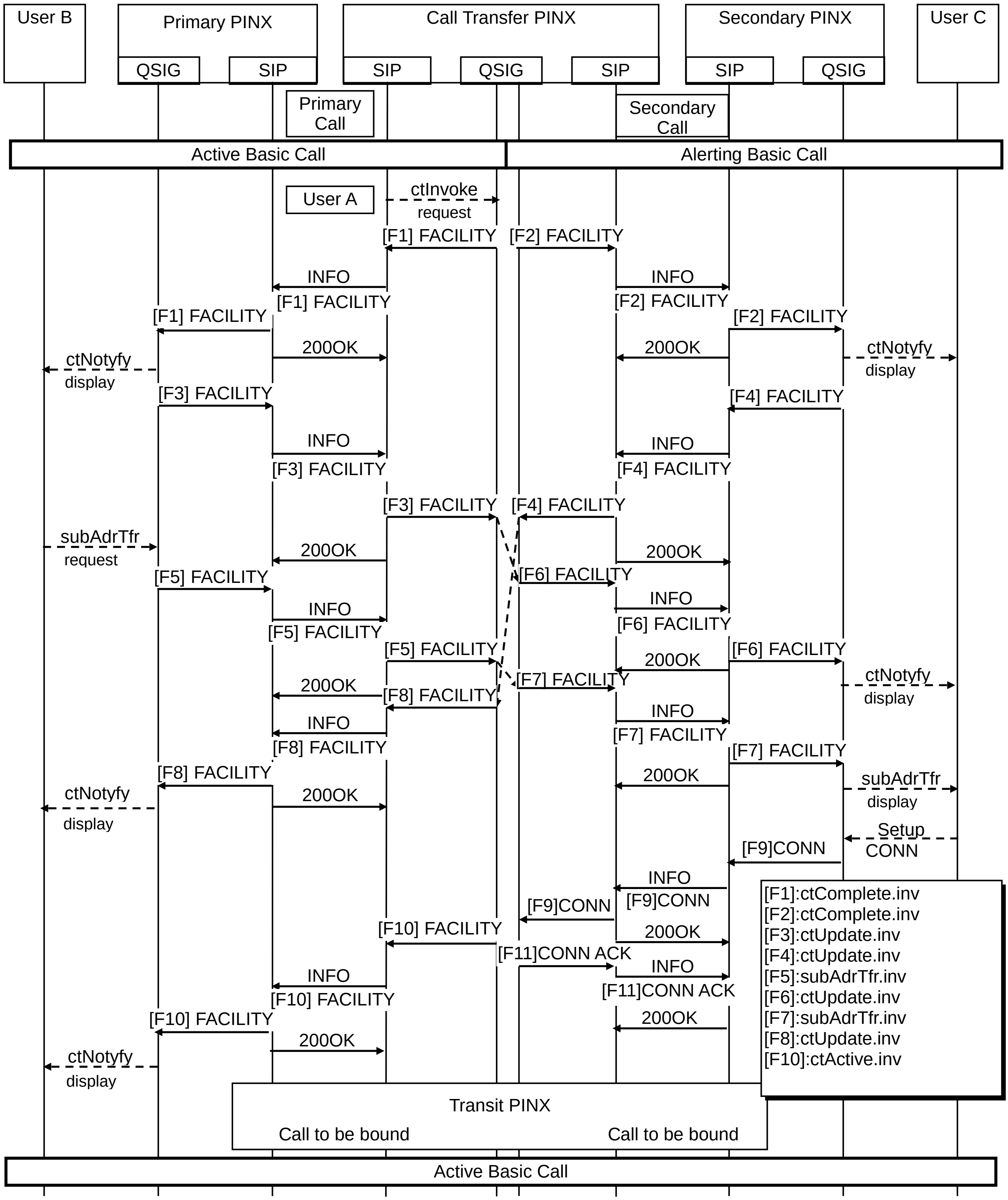
Attached Figure 3.1.3 JJ-22.06 shows an example of normal operation of transit-type call transfer in the case where one call is "Active" and the other is "Alerting".

Note that the sequence example below is a quote from those defined in JS-13869.



Attached Figure 3.1.3 JJ-22.06 Message sequence of normal operation of transit-type SS-CT in the case where one call is "Active" and the other is "Alerting"

(2-1) Example of a message sequence of transit-type call transfer in the case where one call is "Alerting" (in SIP tunneling)

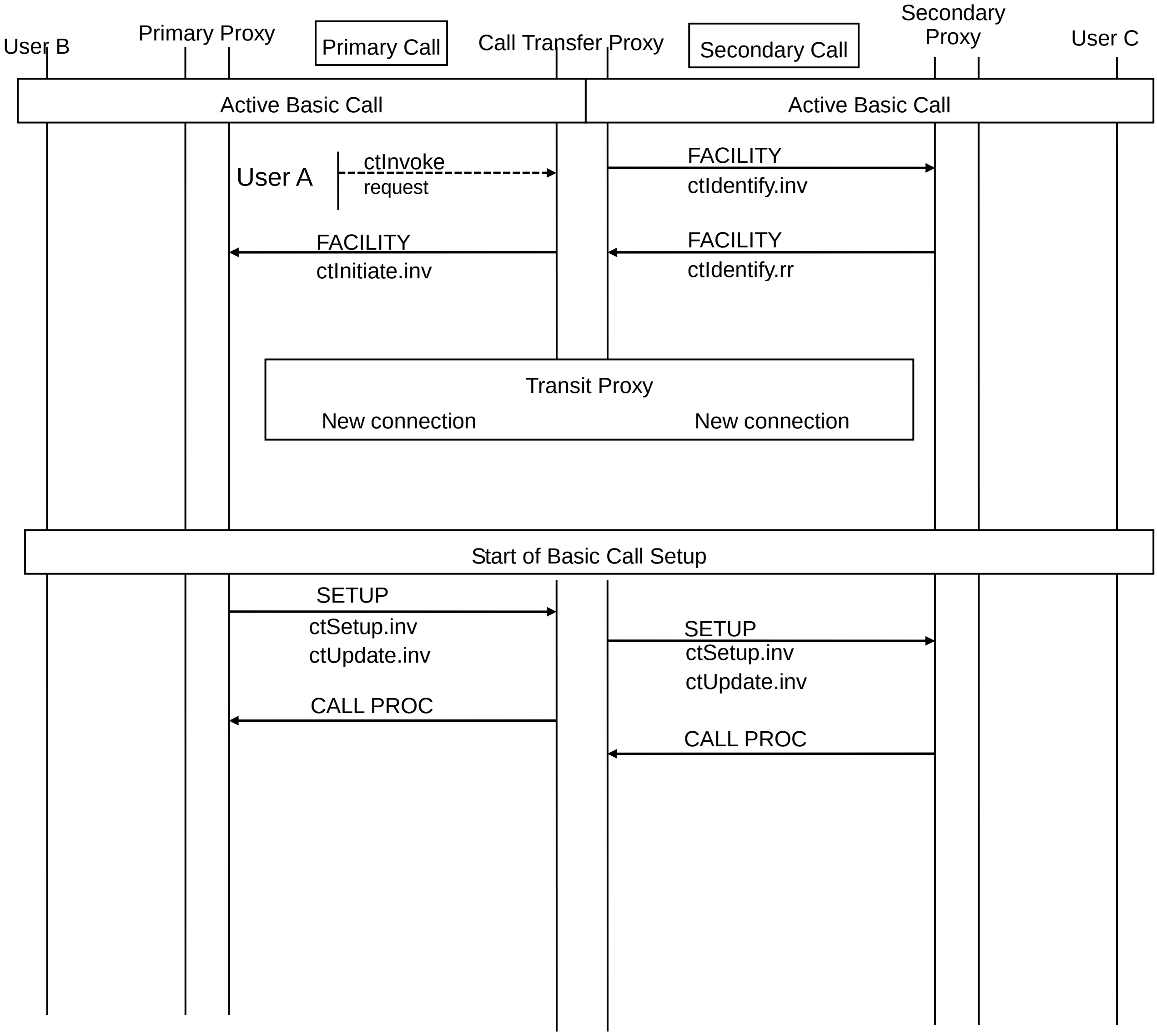


Attached Figure 3.1.4 JJ-22.06 Message sequence of normal operation of transit-type SS-CT in the case where one call is "Active" and the other is "Alerting"

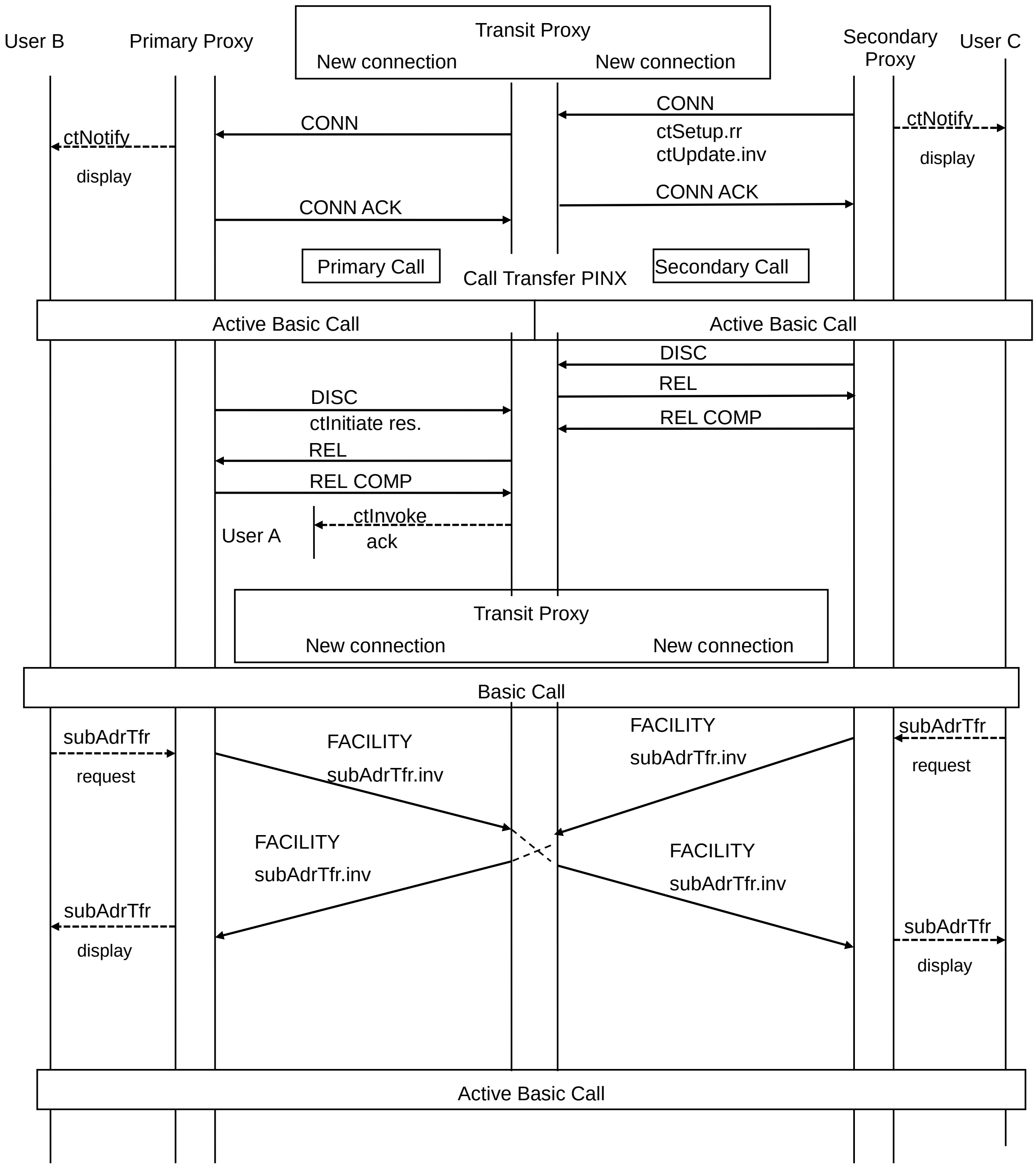
(3) Example of a message sequence of normal operation of rerouting-type call transfer

Attached Figure 3.1.5 JJ-22.06 shows an example of normal operation of rerouting-type call transfer in the case where two calls concerned with call transfer operation are both “Active”.

Note that the sequence example below is a quote from those defined in JS-13869.

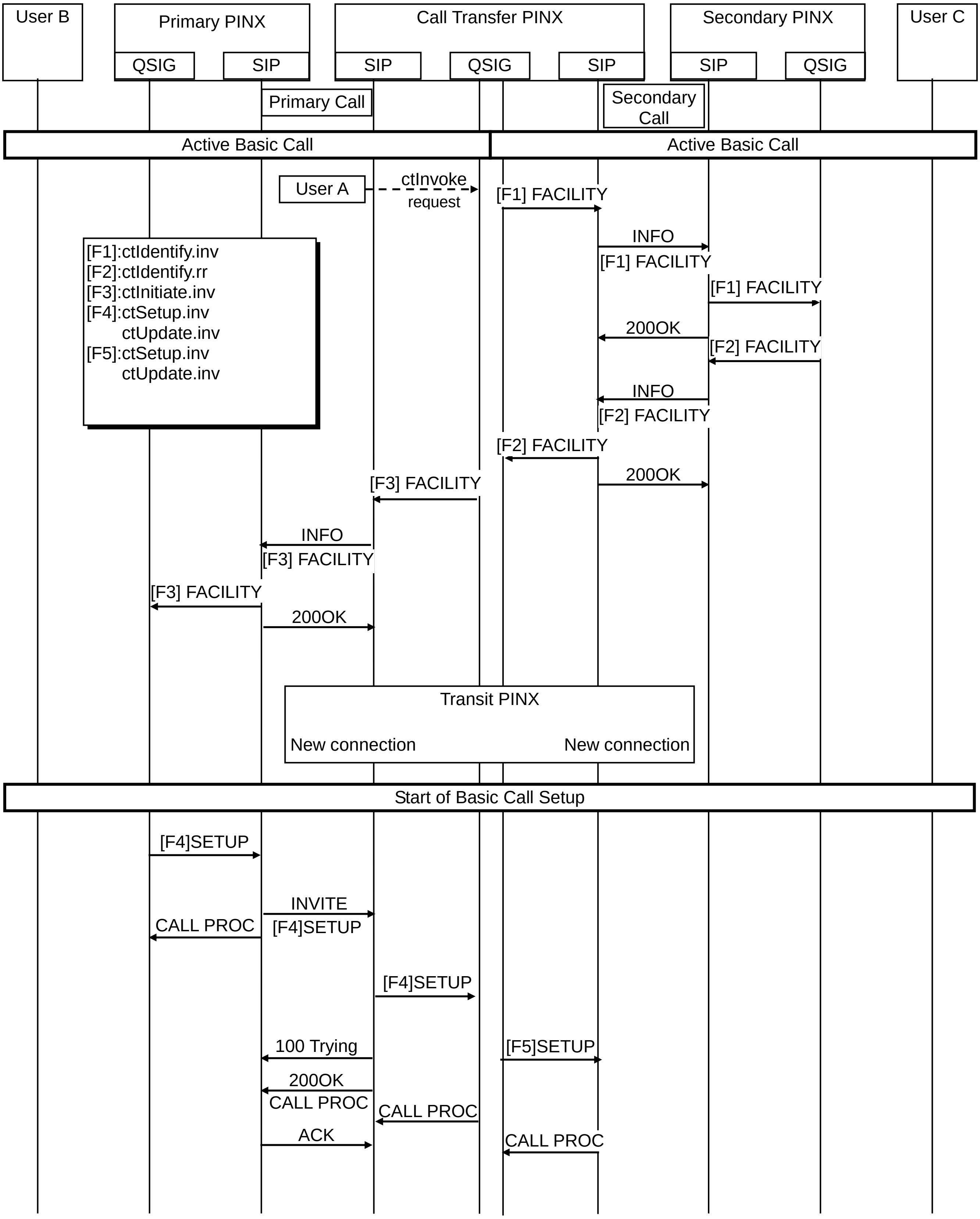


Attached Figure 3.1.5 J-22.06 (1/2) Message sequence of rerouting-type call transfer in the case where both calls are “Active”

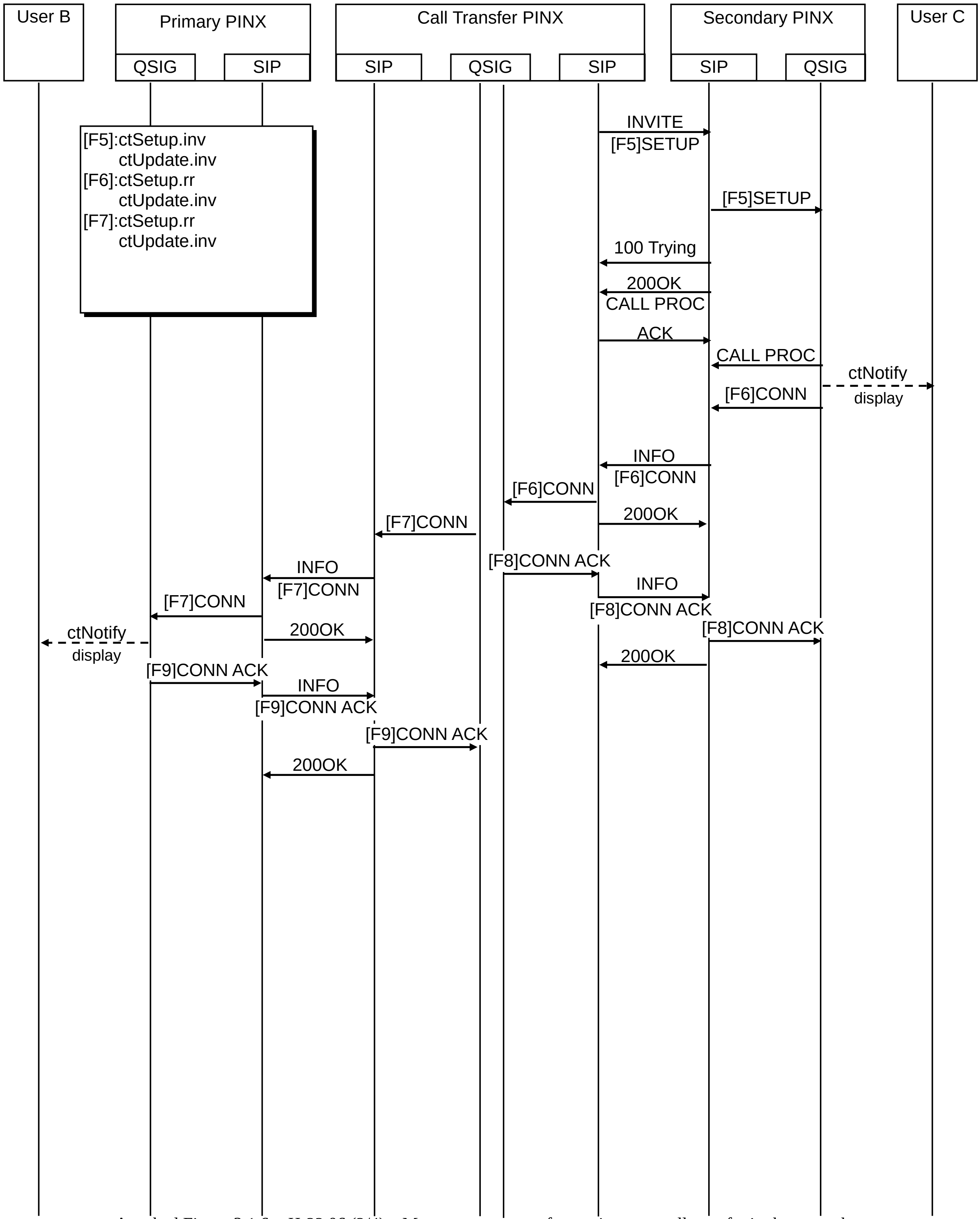


Attached Figure 3.1.5 JJ-22.06 (2/2) Message sequence of rerouting-type call transfer in the case where both calls are "Active"

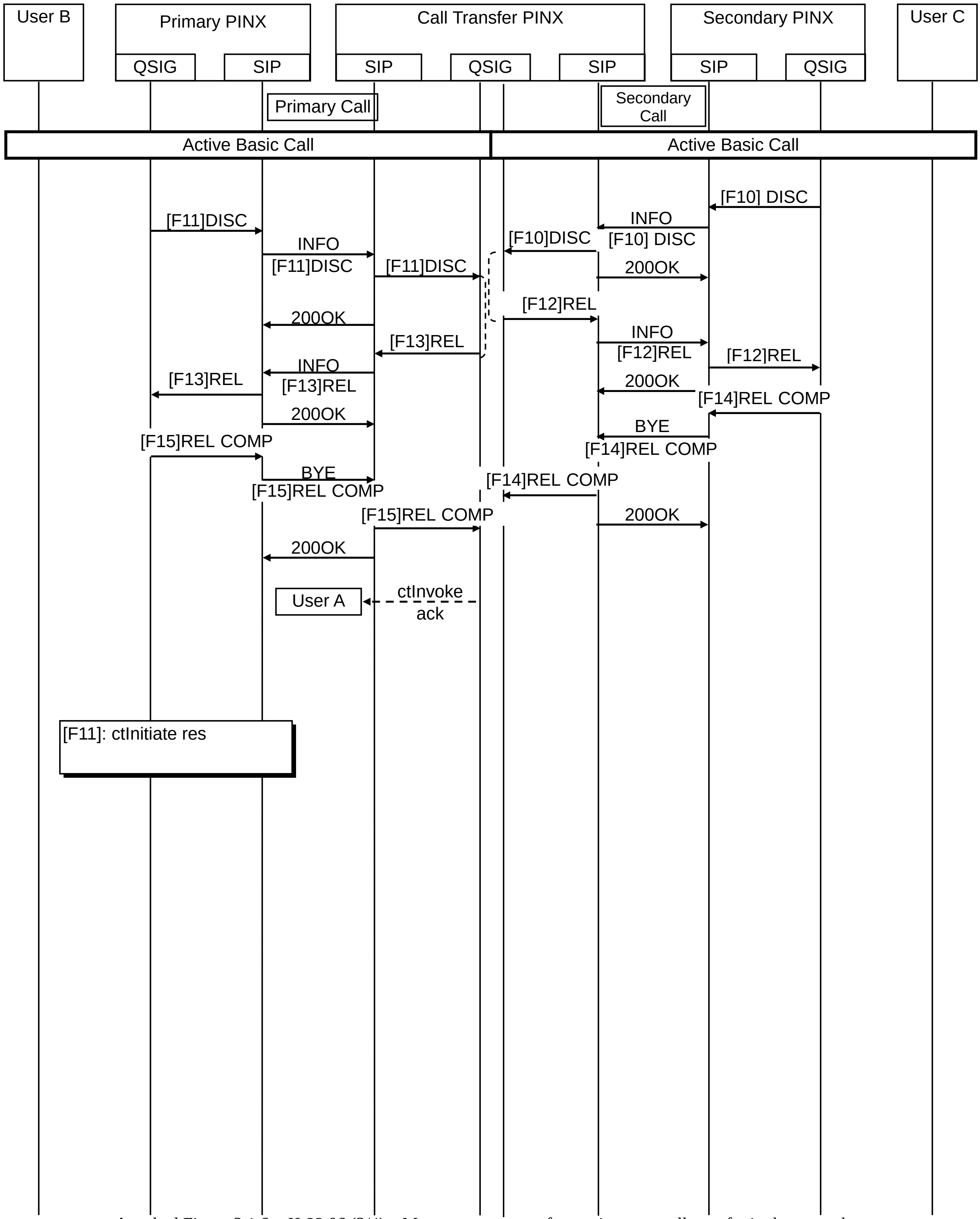
(3-1) Example of a message sequence of normal operation of rerouting-type call transfer **(in SIP tunneling)**



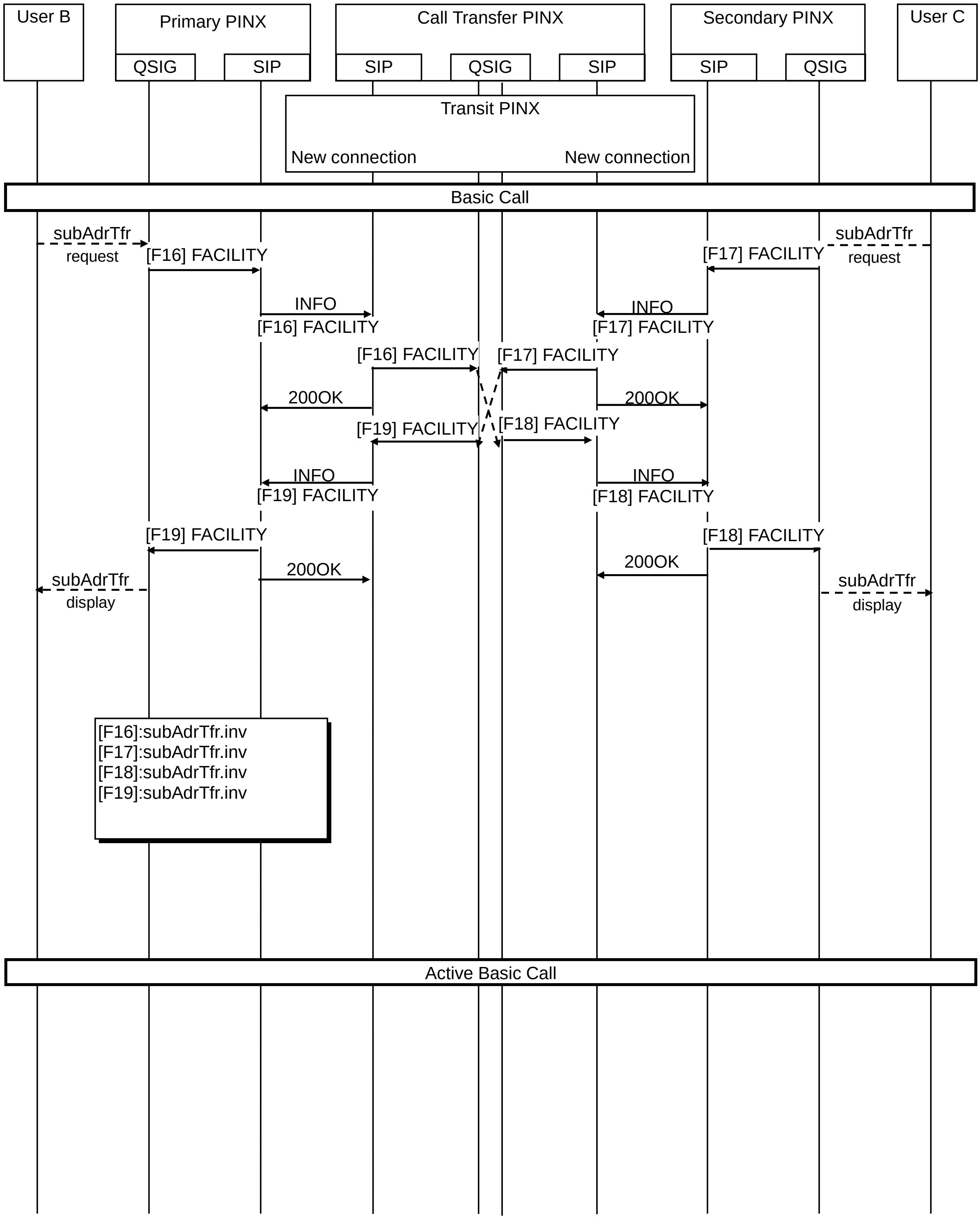
Attached Figure 3.1.6 JJ-22.06 (1/4) Message sequence of rerouting-type call transfer in the case where both calls are "Active"



Attached Figure 3.1.6 JJ-22.06 (2/4) Message sequence of rerouting-type call transfer in the case where both calls are “Active”



Attached Figure 3.1.6 JJ-22.06 (3/4) Message sequence of rerouting-type call transfer in the case where both calls are “Active”

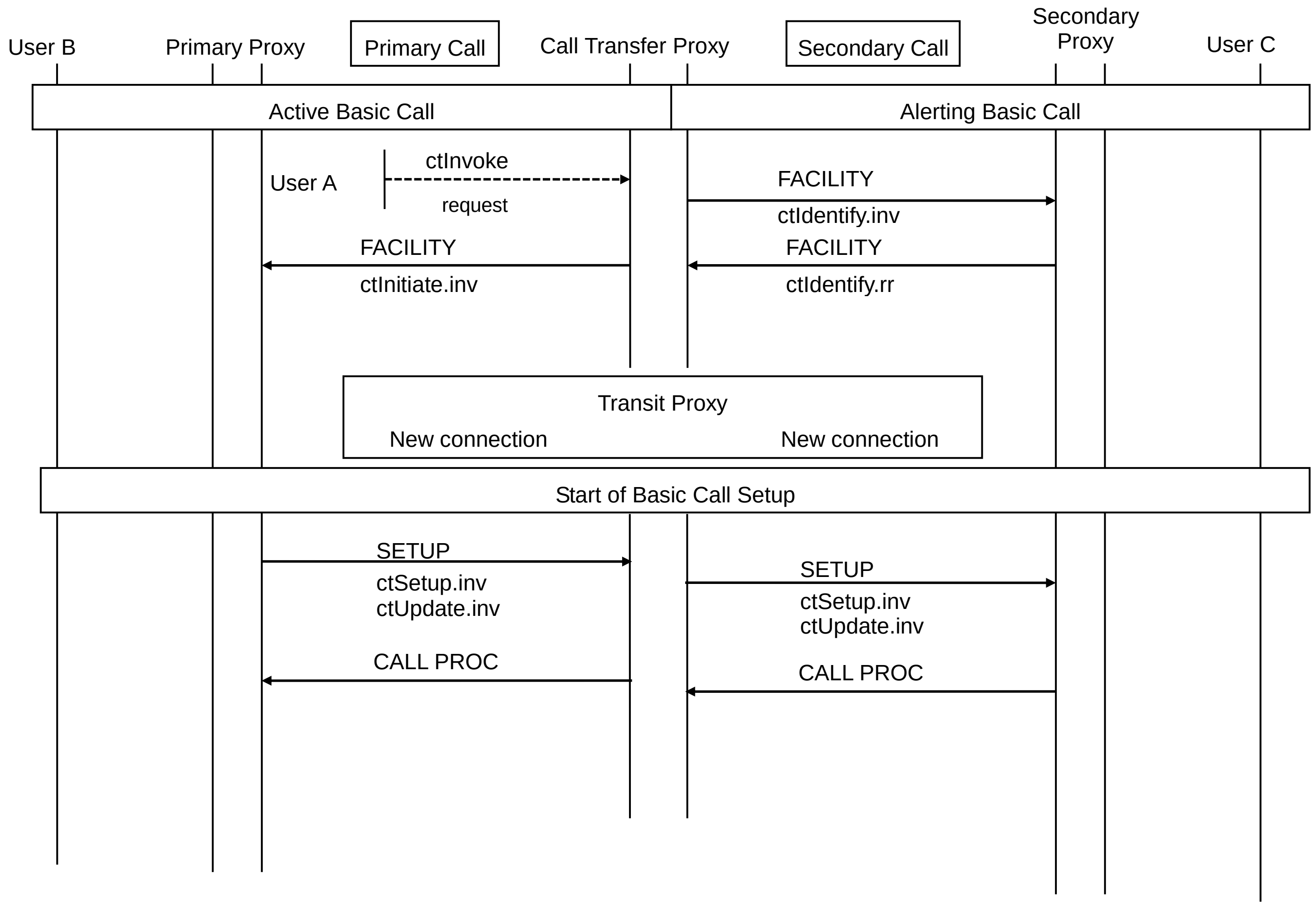


Attached Figure 3.1.6 JJ-22.06 (4/4) Message sequence of rerouting-type call transfer in the case where both calls are "Active"

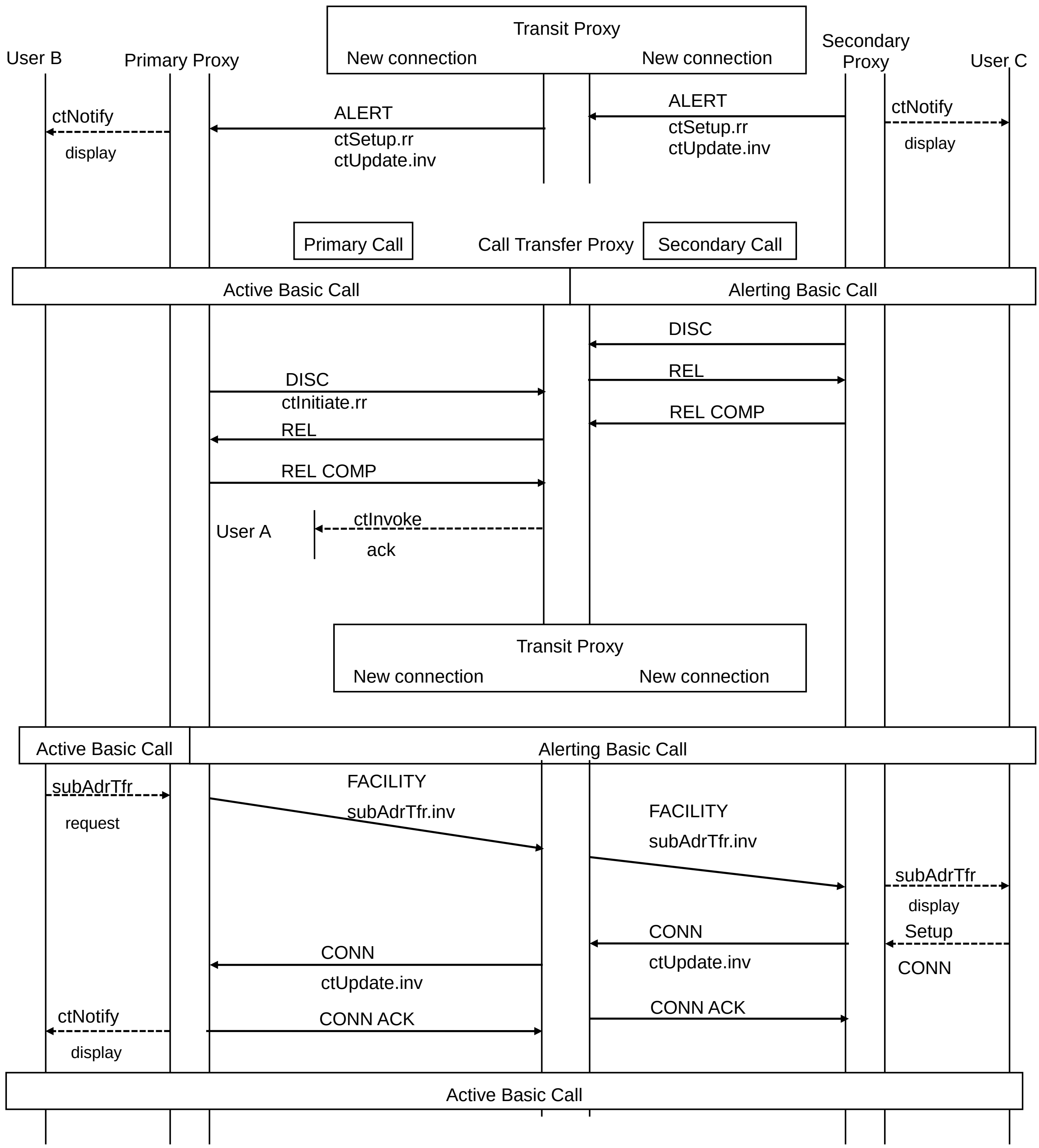
(4) Example of a message sequence of normal operation of rerouting-type call transfer in the case where one call is "Alerting"

Attached Figure 3.1.7 JJ-22.06 shows an example of normal operation of rerouting-type call transfer in the case where one call is "Active" and the other is "Alerting".

Note that the sequence example below is a quote from those defined in JS-13869.

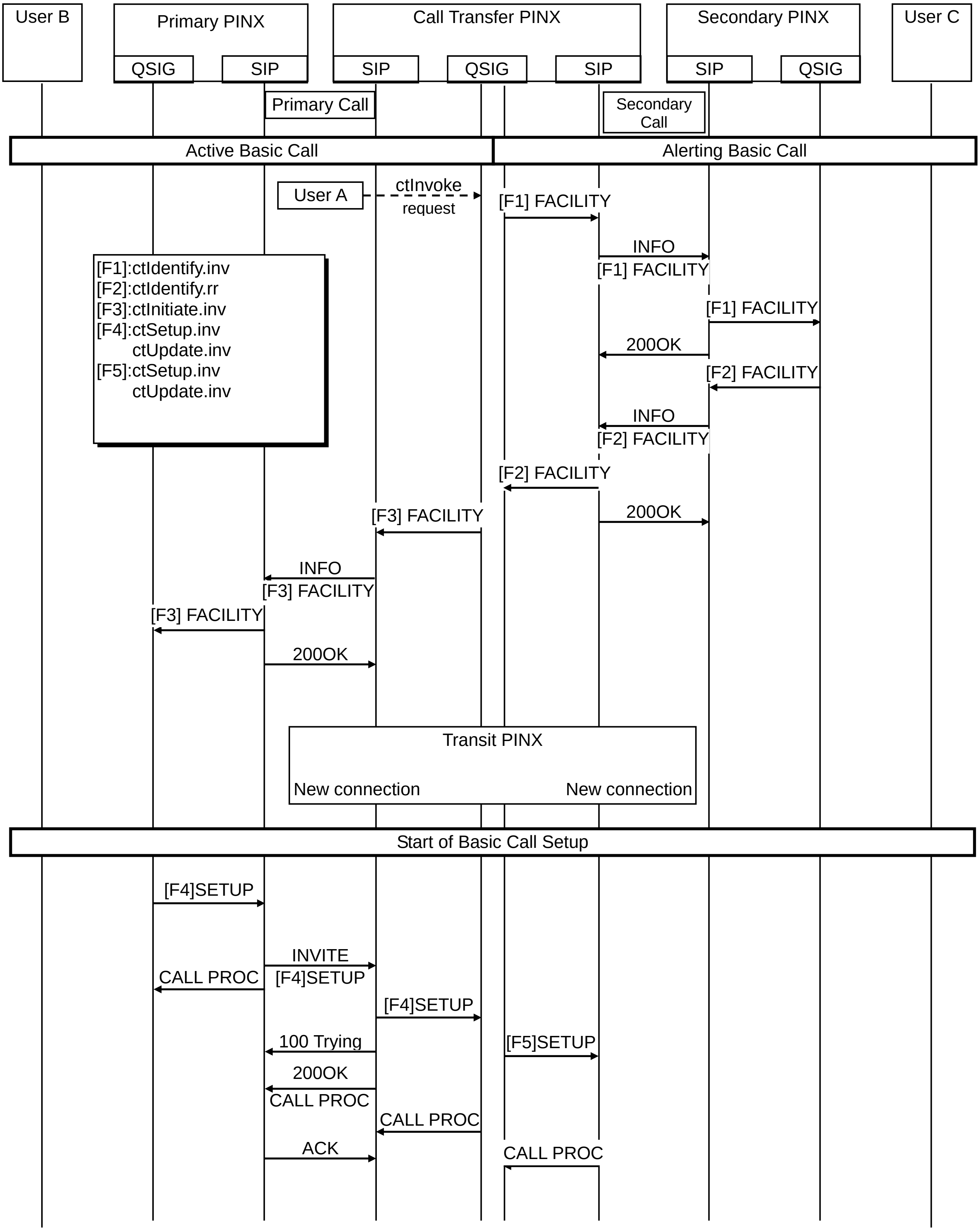


Attached Figure 3.1.7 JJ-22.06 (1/2) Message sequence of rerouting-type call transfer in the case where one call is "Active" and the other is "Alerting"

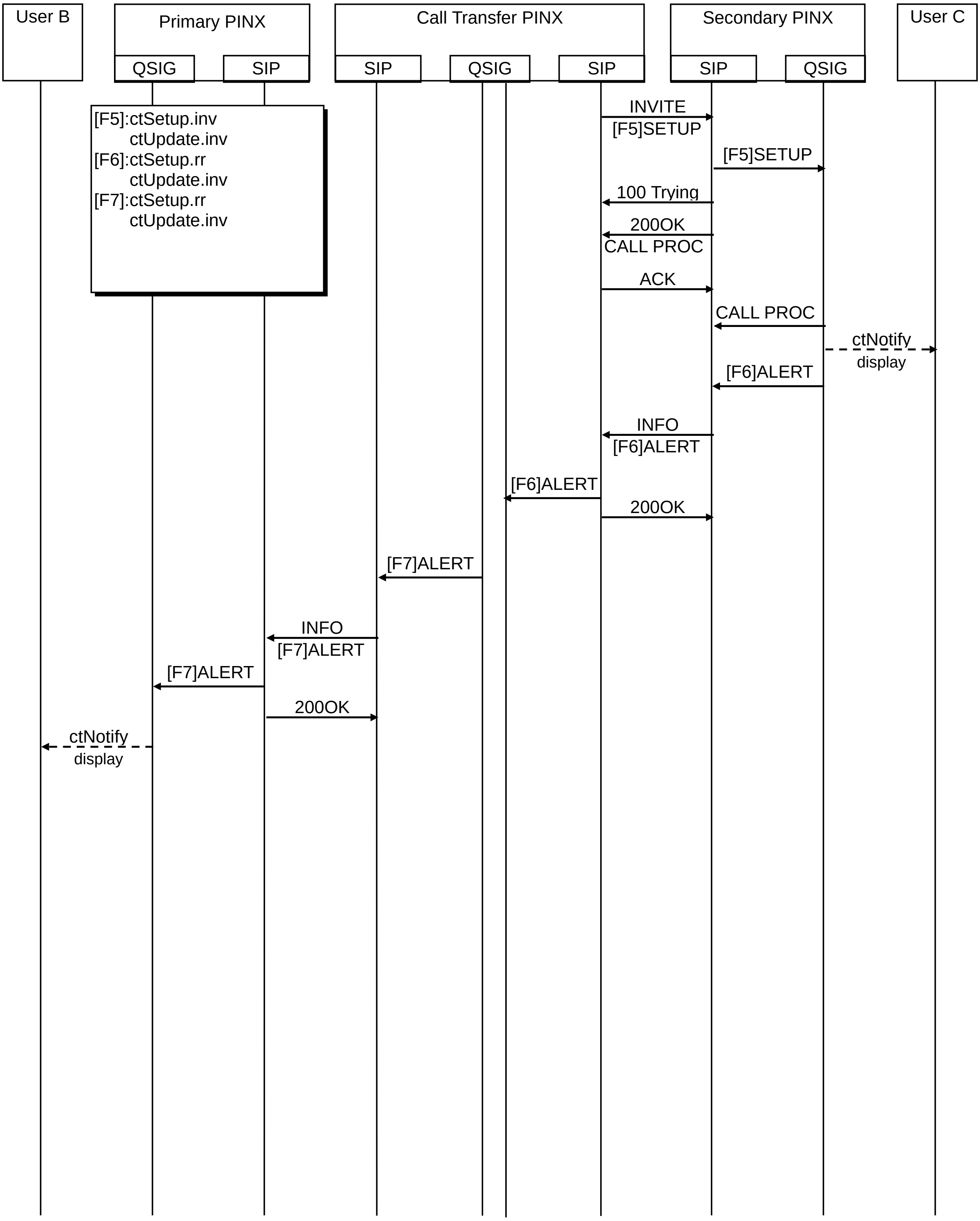


Attached Figure 3.1.7 JJ-22.06 (2/2) Message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”

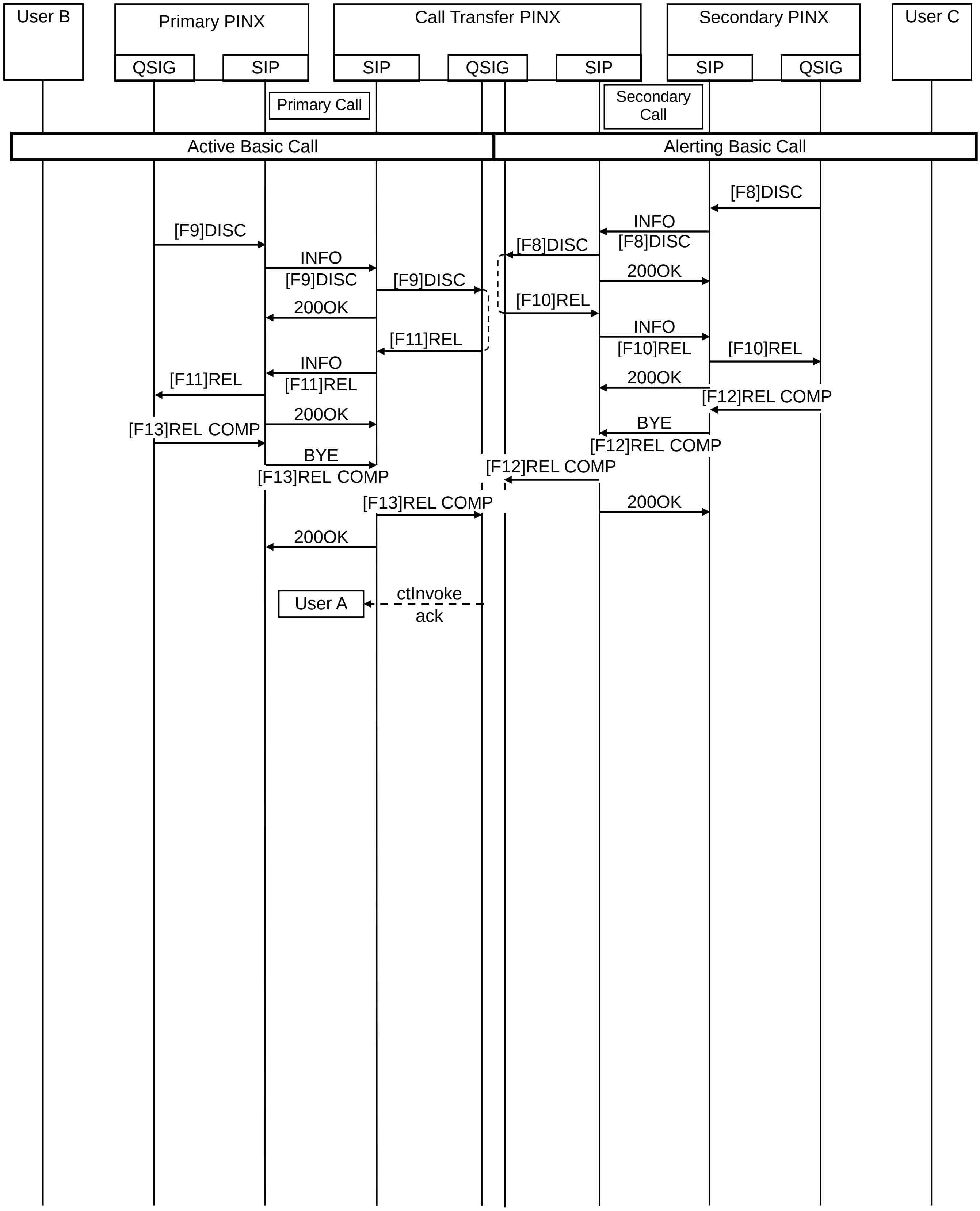
(4-1) Example of a message sequence of normal operation of rerouting-type call transfer in the case where one call is "Alerting" **(in SIP tunneling)**



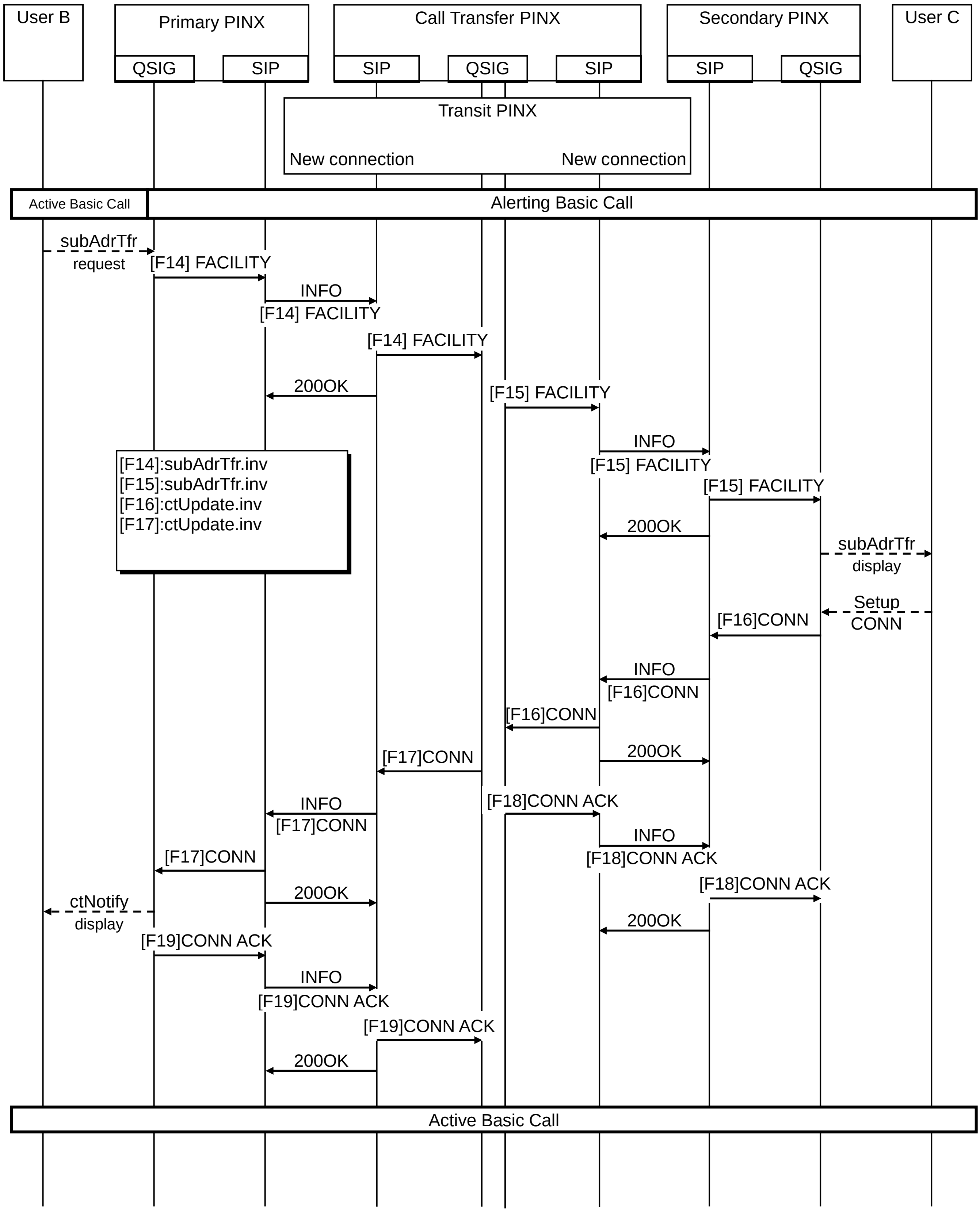
Attached Figure 3.1.8 JJ-22.06 (1/4) Message sequence of rerouting-type call transfer in the case where one call is "Active" and the other is "Alerting" (tunneling)



Attached Figure 3.1.8 JJ-22.06 (2/4) Message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting” (tunneling)



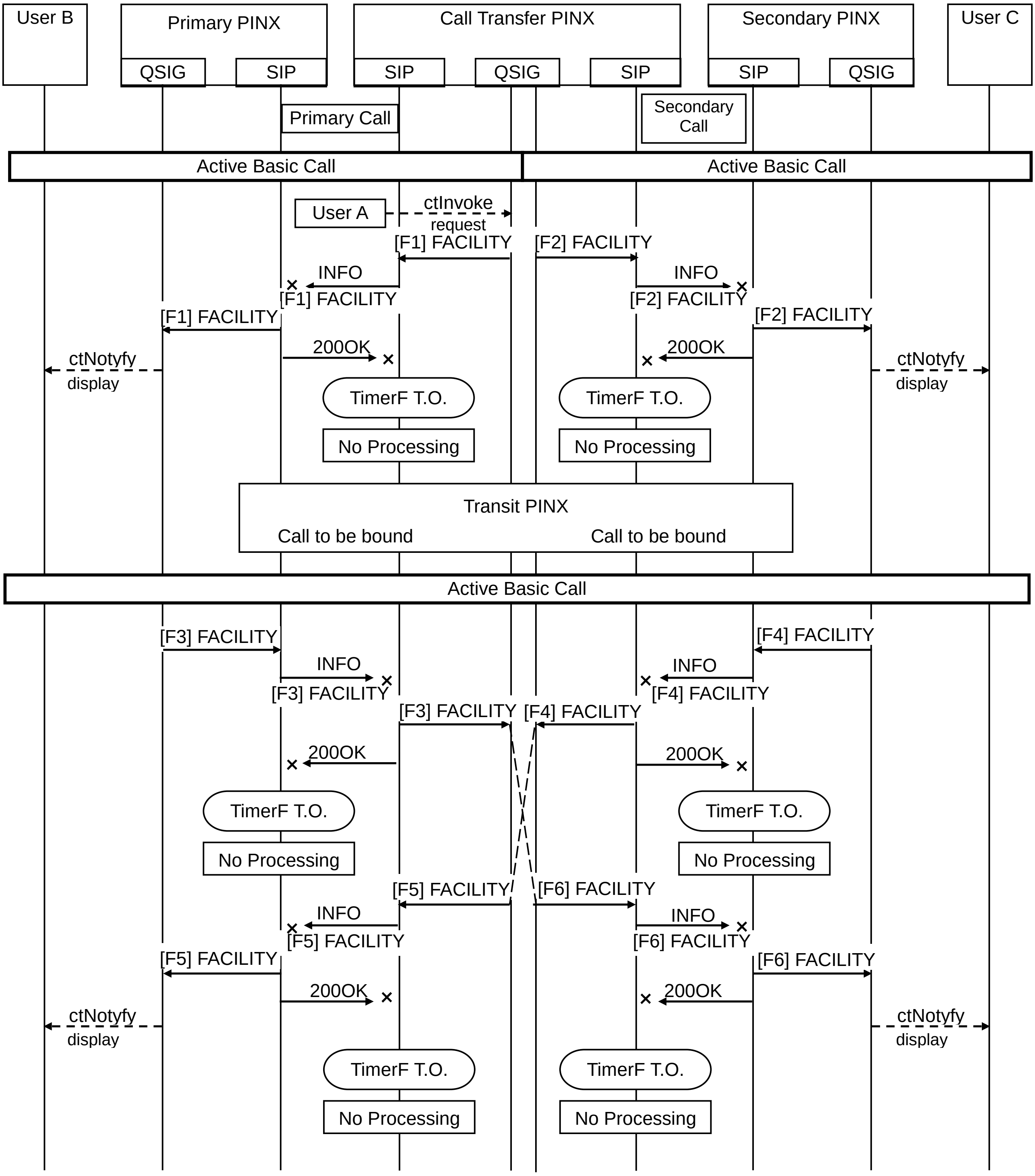
Attached Figure 3.1.8 JJ-22.06 (3/4) Message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting” (tunneling)



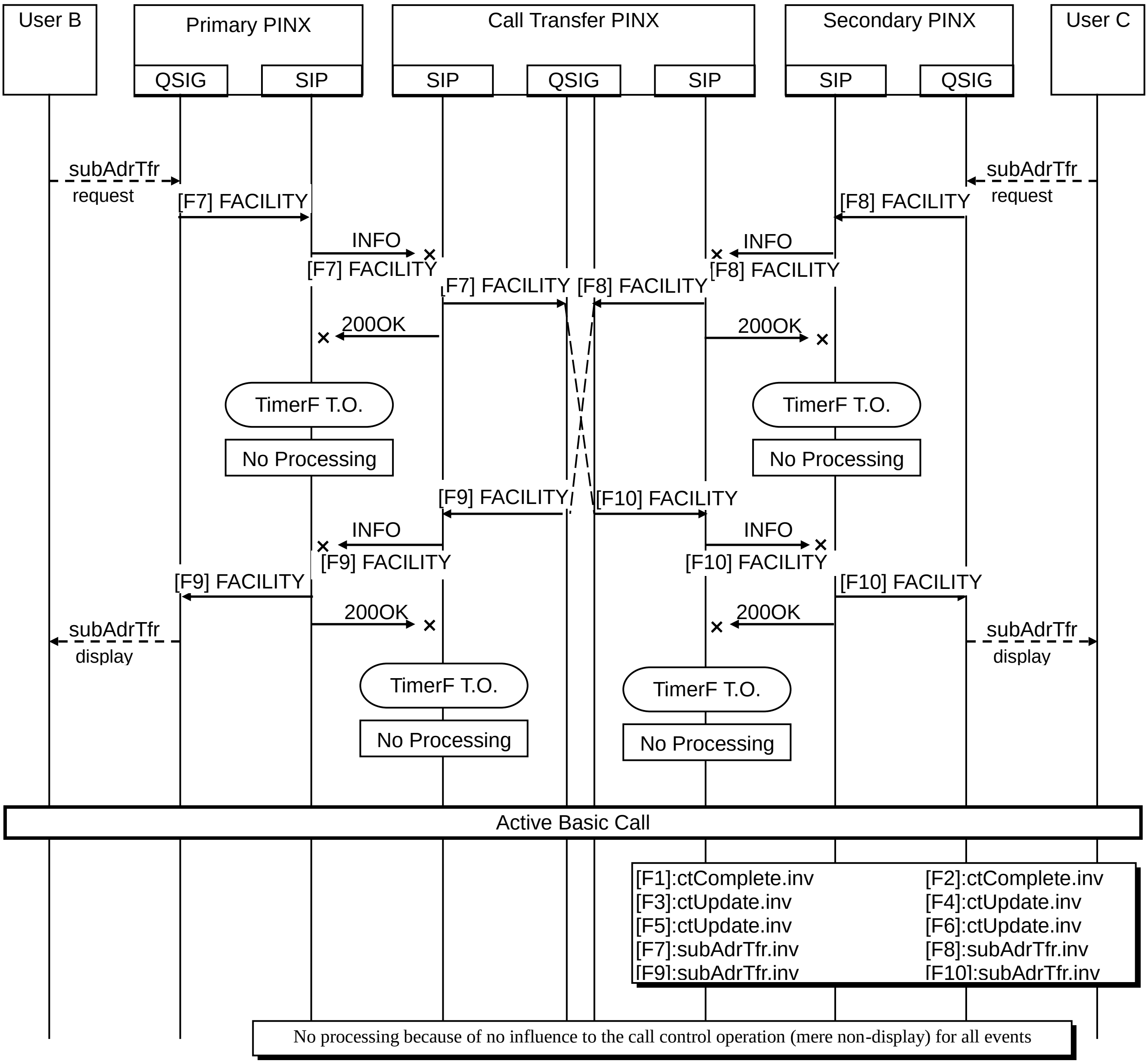
Attached Figure 3.1.8 JJ-22.06 (4/4) Message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting” (tunneling)

3.2 Procedures (quasi-normal sequence)

(1) Example of a message sequence of quasi-normal operation of transit-type call transfer in the case where both calls are "Active" (in SIP tunneling)

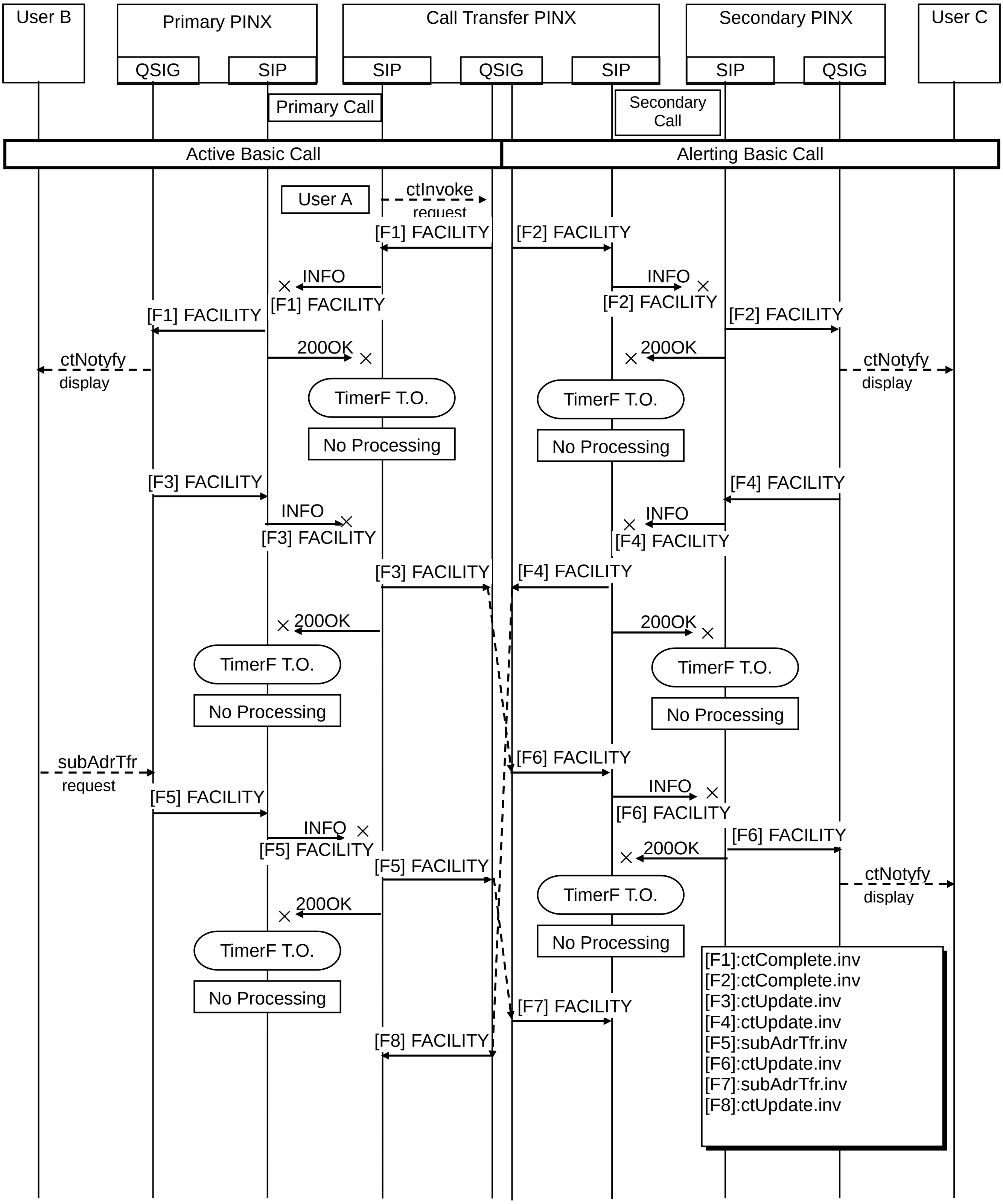


Attached Figure 3.2.1 JJ-22.06 (1/2) Quasi-normal message sequence of transit-type SS-CT operation in the case where both calls are "Active"

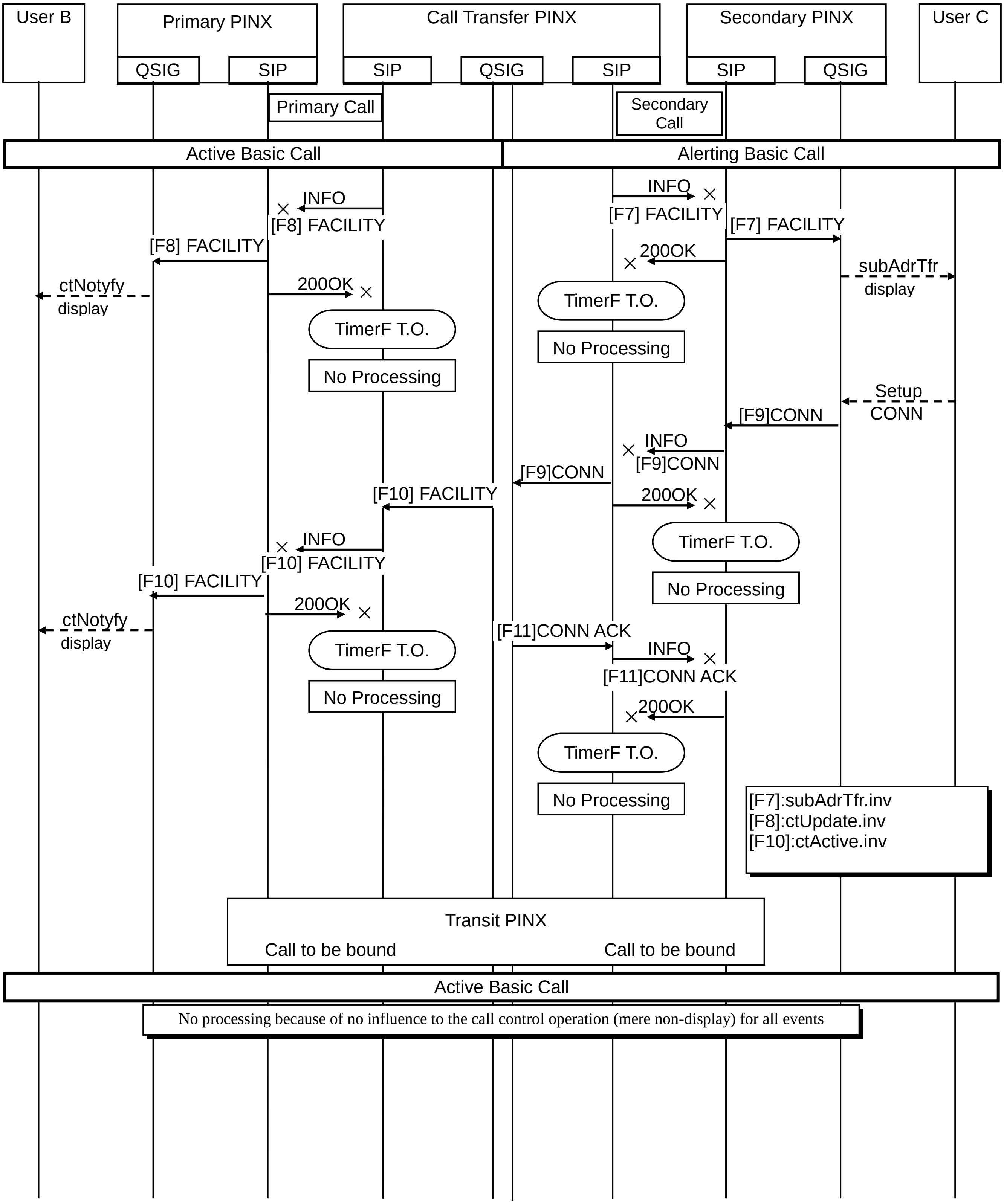


Attached Figure 3.2.1 JJ-22.06 (2/2) Quasi-normal message sequence of transit-type SS-CT operation in the case where both calls are "Active"

(2) Example of a message sequence of quasi-normal operation of transit-type call transfer in the case where one call is "Alerting" **(in SIP tunneling)**

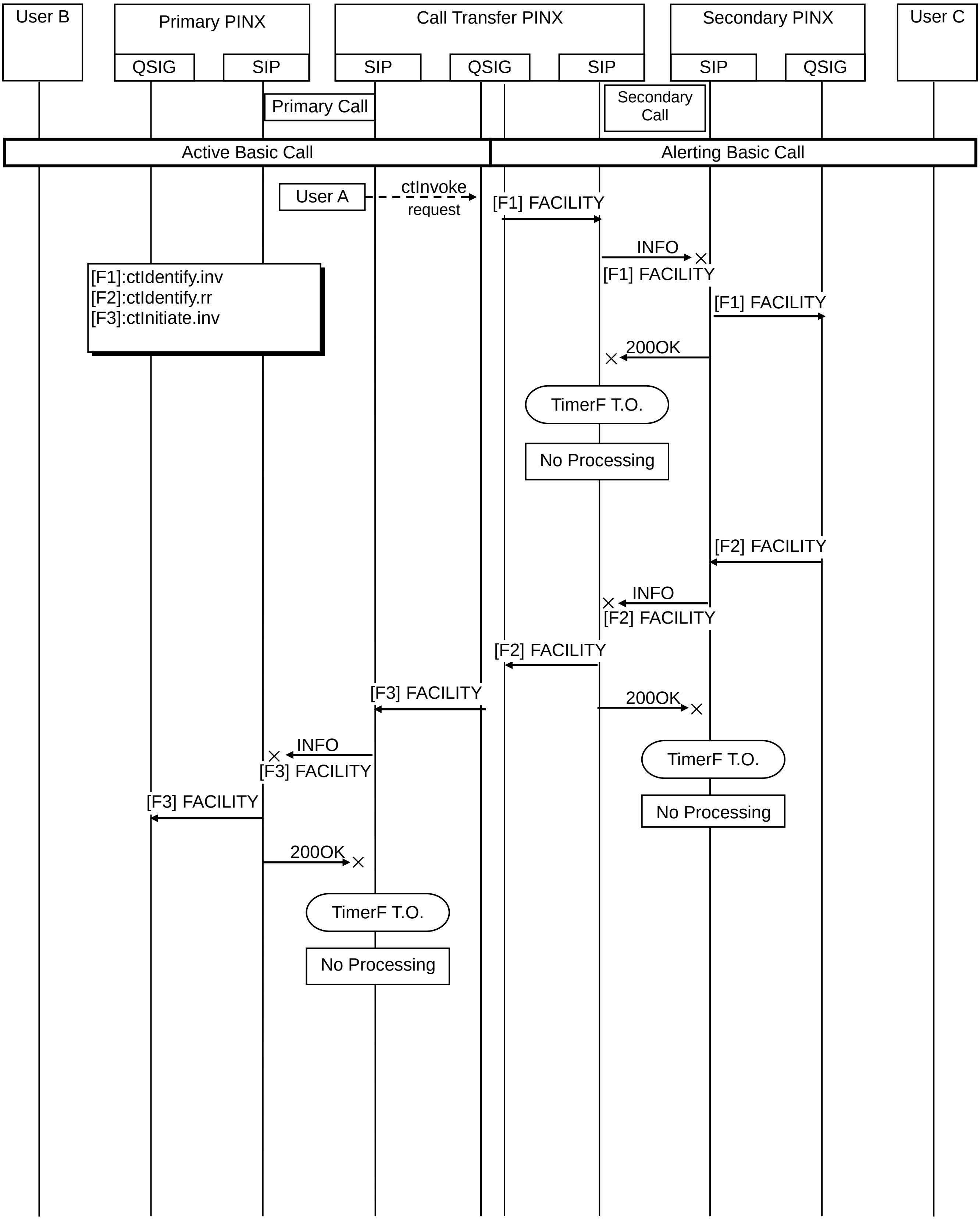


Attached Figure 3.2.2 JJ-22.06 (1/2) Quasi-normal message sequence of transmit-type SS-CT operation in the case where one call is "Active" and the other is "Alerting"

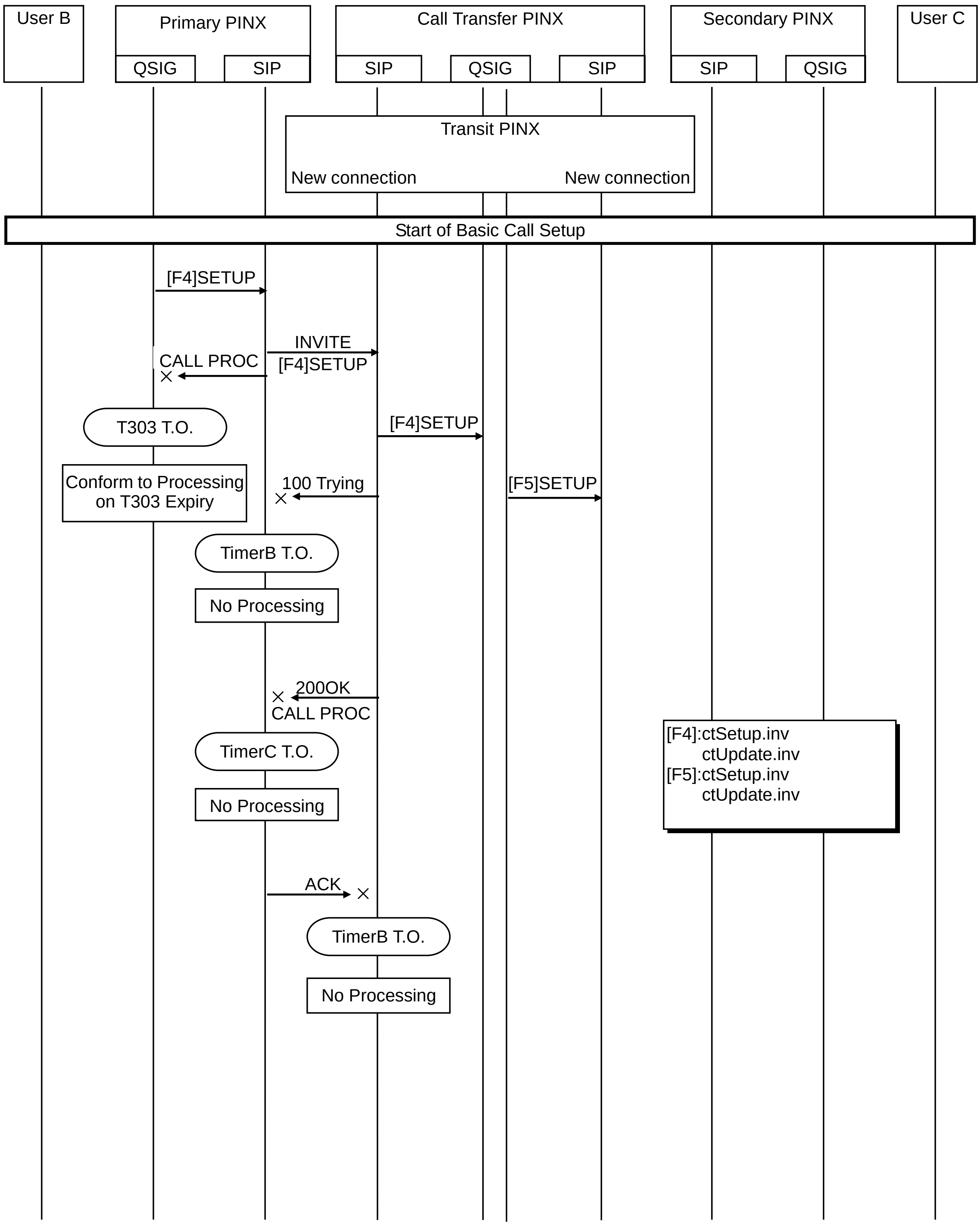


Attached Figure 3.2.2 JJ-22.06 (2/2) Quasi-normal message sequence of transmit-type SS-CT operation in the case where one call is “Active” and the other is “Alerting”

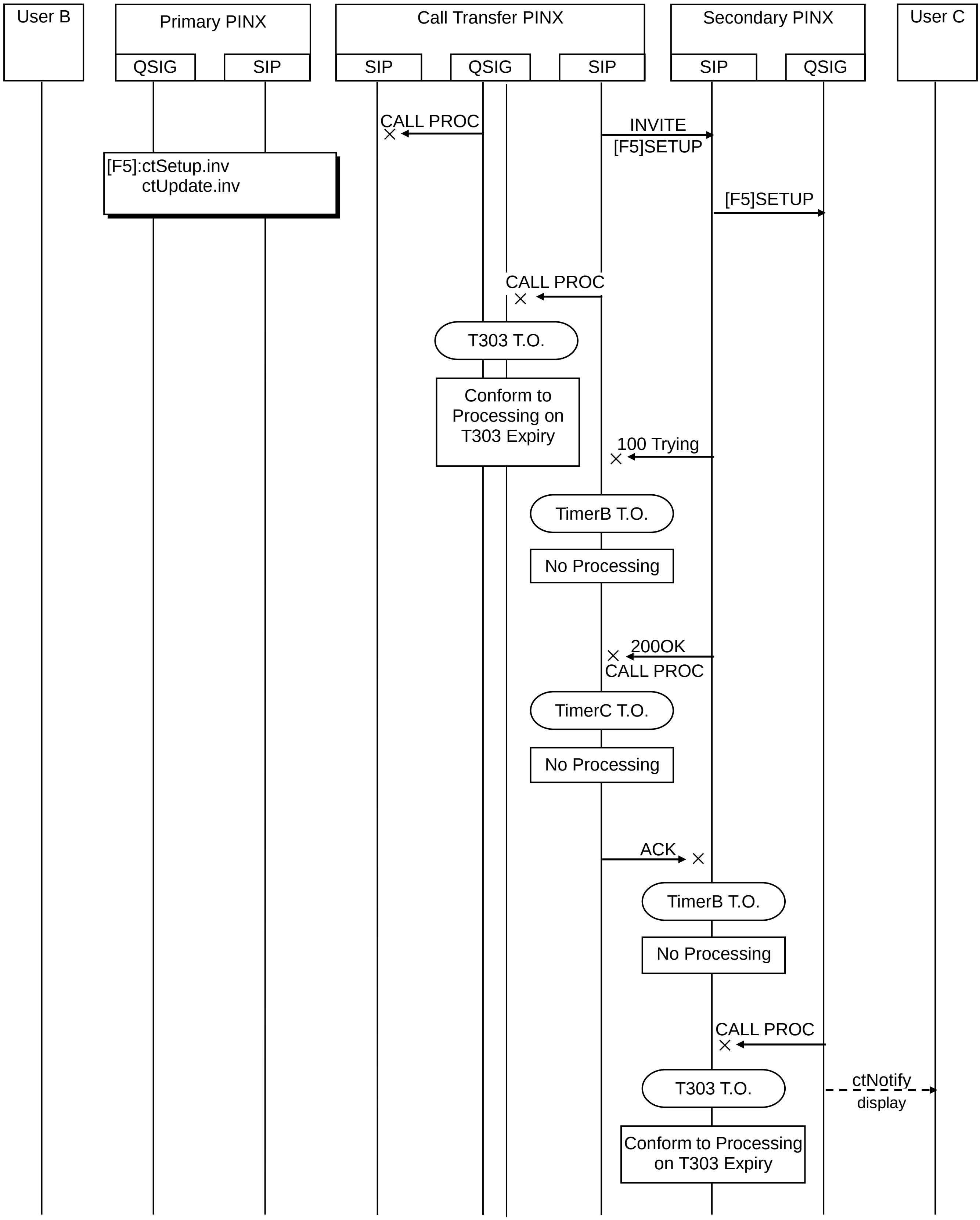
(3) Example of a message sequence of quasi-normal operation of rerouting-type call transfer (in SIP tunneling)



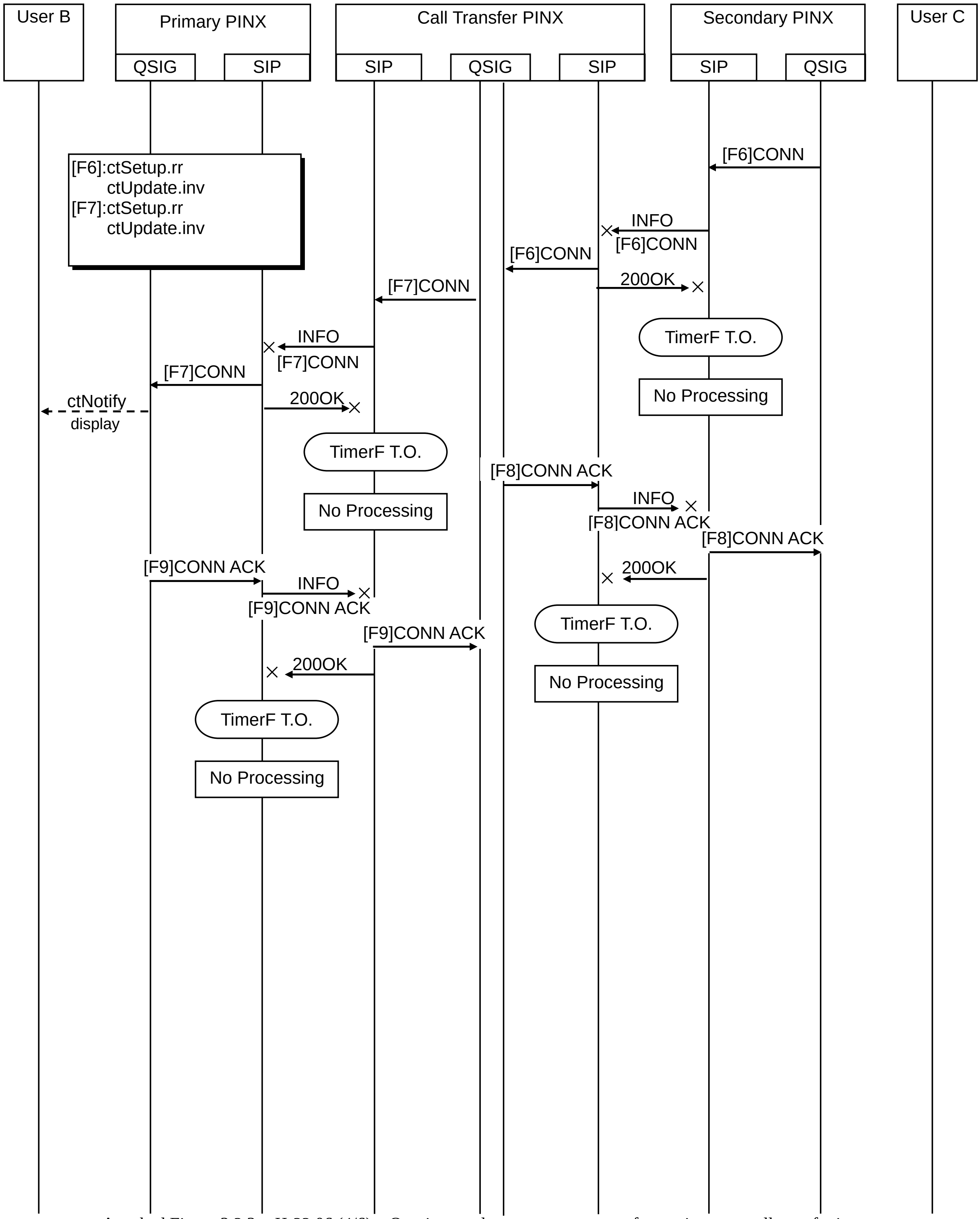
Attached Figure 3.2.3 JJ-22.06 (1/6) Quasi-normal message sequence of rerouting-type call transfer in the case where both calls are “Active”



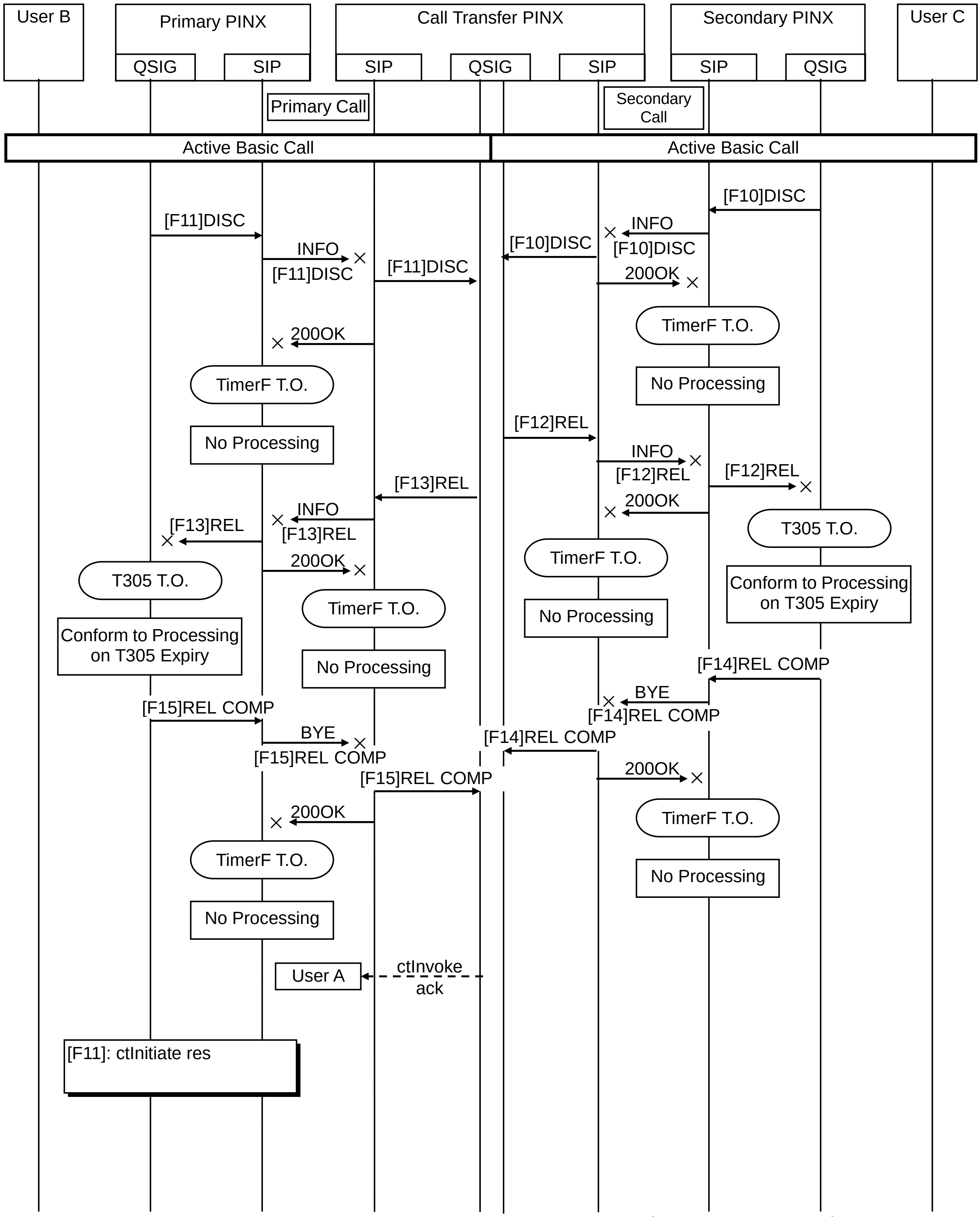
Attached Figure 3.2.3 JJ-22.06 (2/6) Quasi-normal message sequence of rerouting-type call transfer in the case where both calls are “Active”



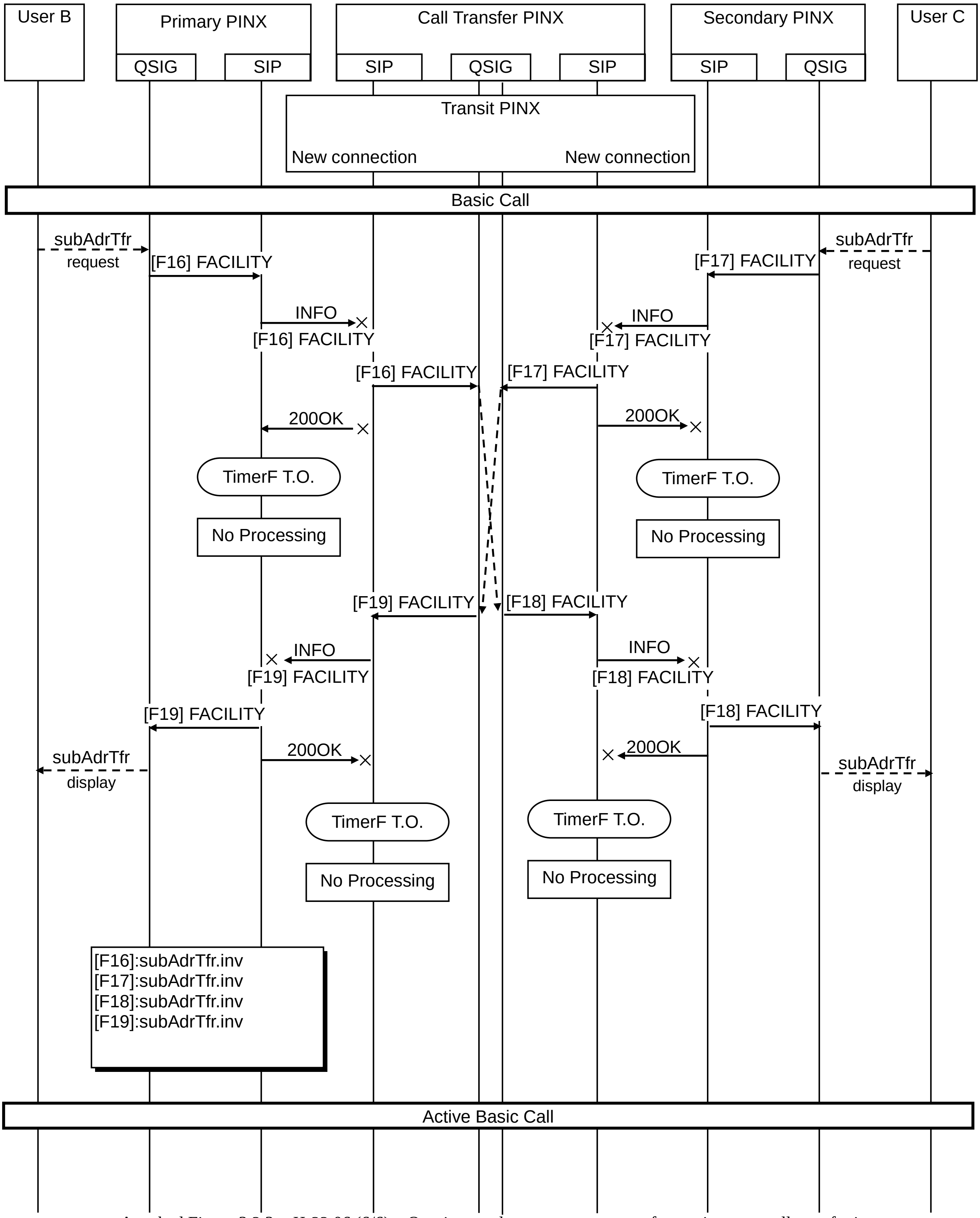
Attached Figure 3.2.3 JJ-22.06 (3/6) Quasi-normal message sequence of rerouting-type call transfer in the case where both calls are “Active”



Attached Figure 3.2.3 JJ-22.06 (4/6) Quasi-normal message sequence of rerouting-type call transfer in the case where both calls are “Active”

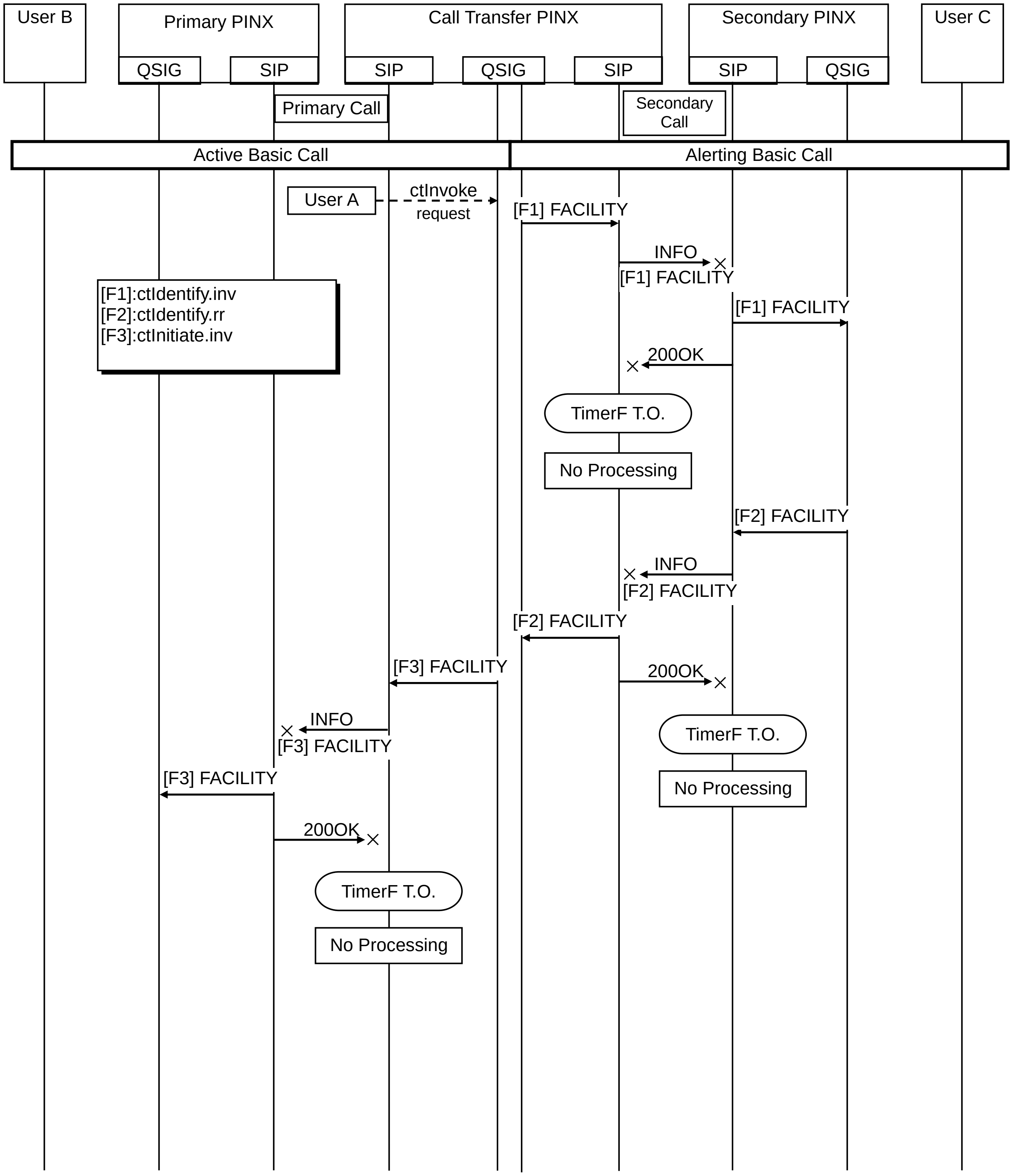


Attached Figure 3.2.3 JJ-22.06 (5/6) Quasi-normal message sequence of rerouting-type call transfer in the case where both calls are "Active"

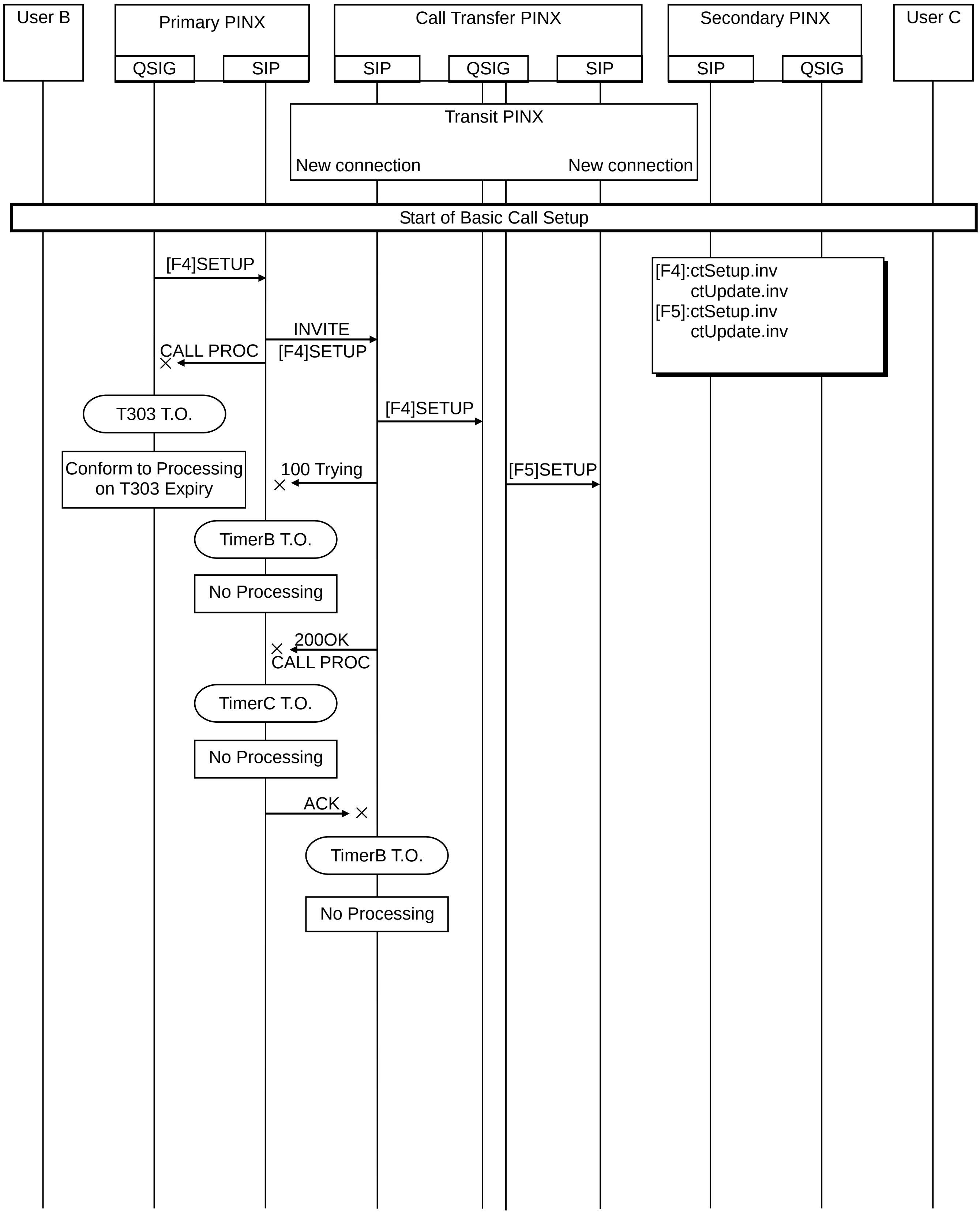


Attached Figure 3.2.3 JJ-22.06 (6/6) Quasi-normal message sequence of rerouting-type call transfer in the case where both calls are “Active”

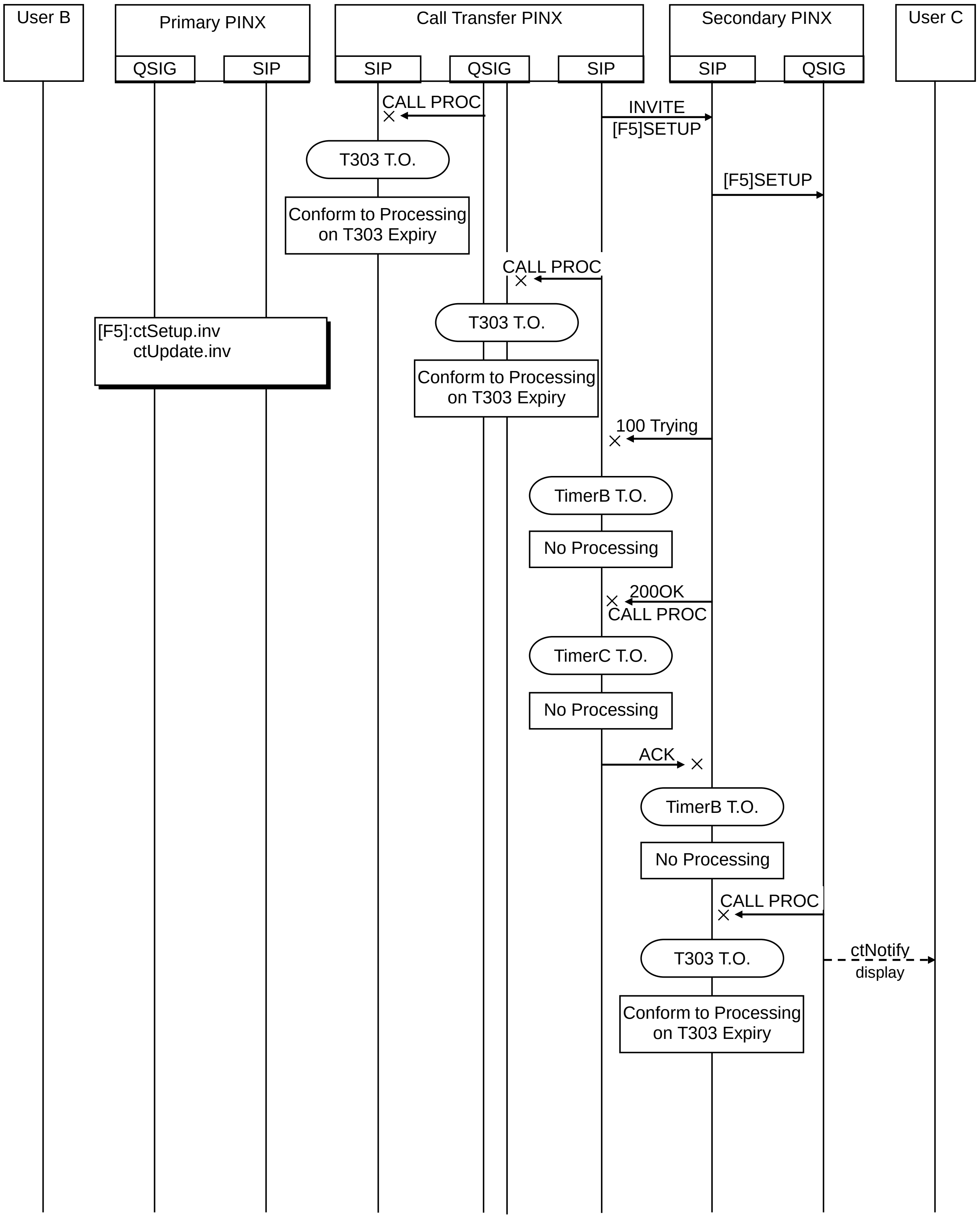
(4) Example of a message sequence of quasi-normal operation of rerouting-type call transfer in the case where one call is “Alerting” (in SIP tunneling)



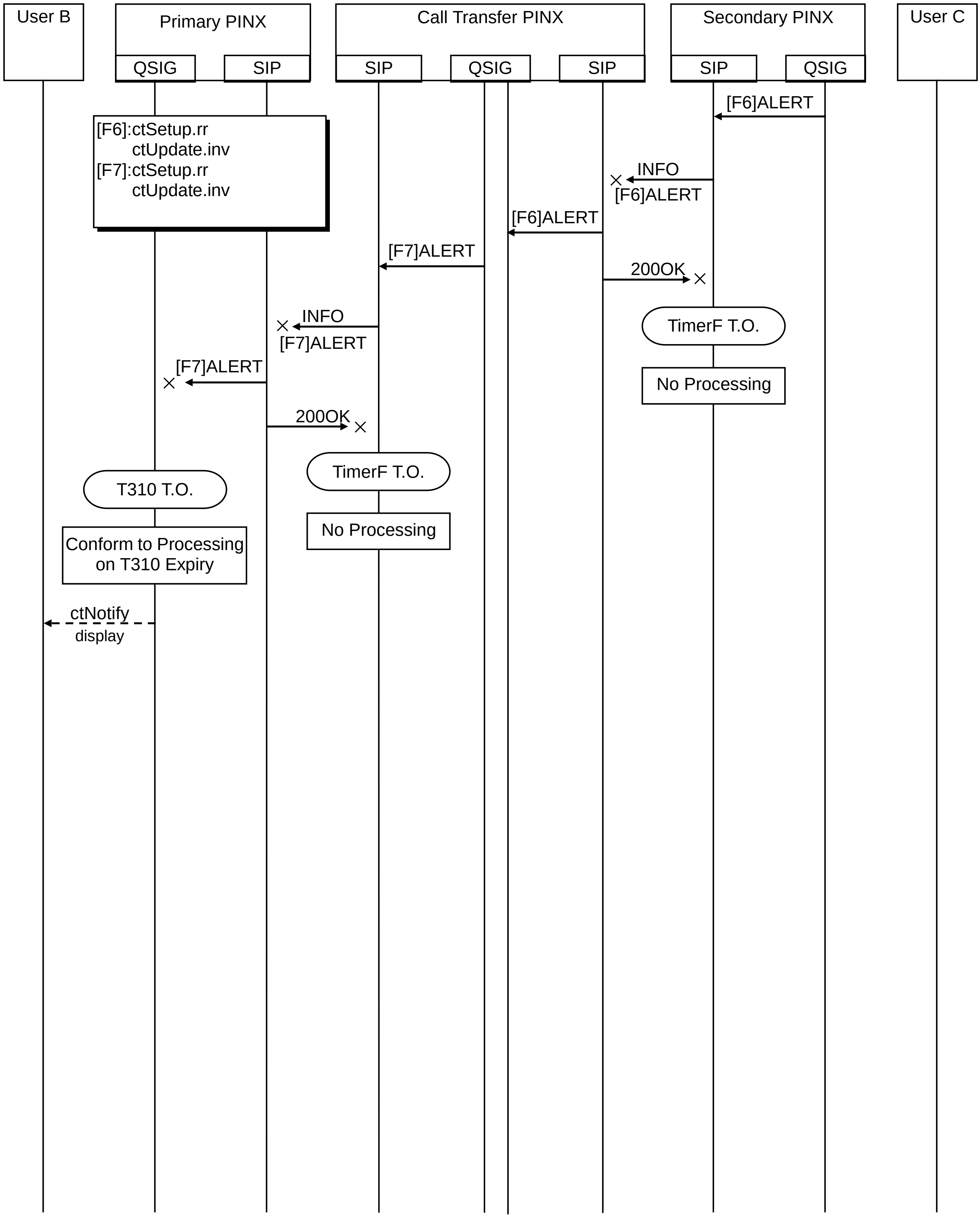
Attached Figure 3.2.4 JJ-22.06 (1/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”



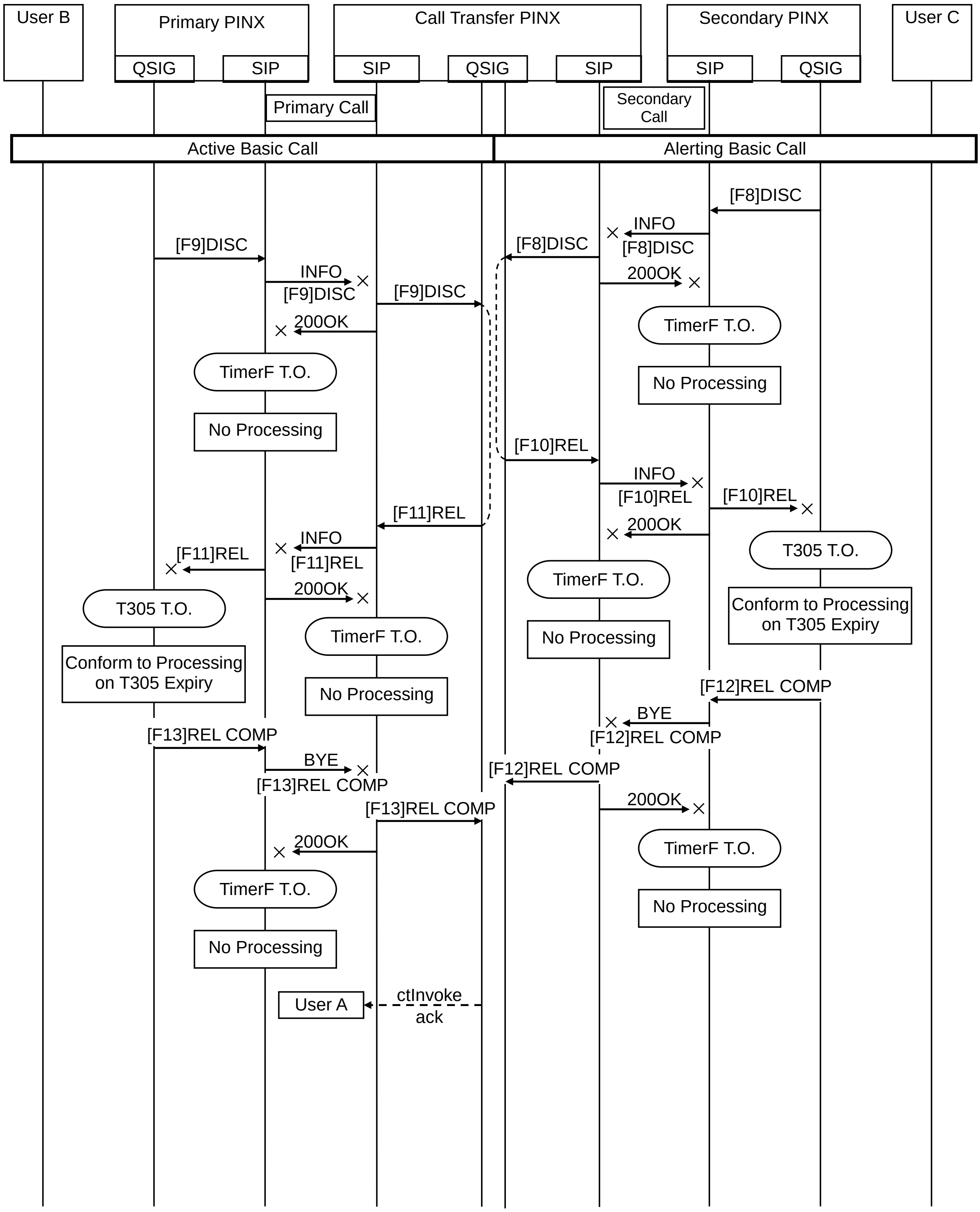
Attached Figure 3.2.4 JJ-22.06 (2/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”



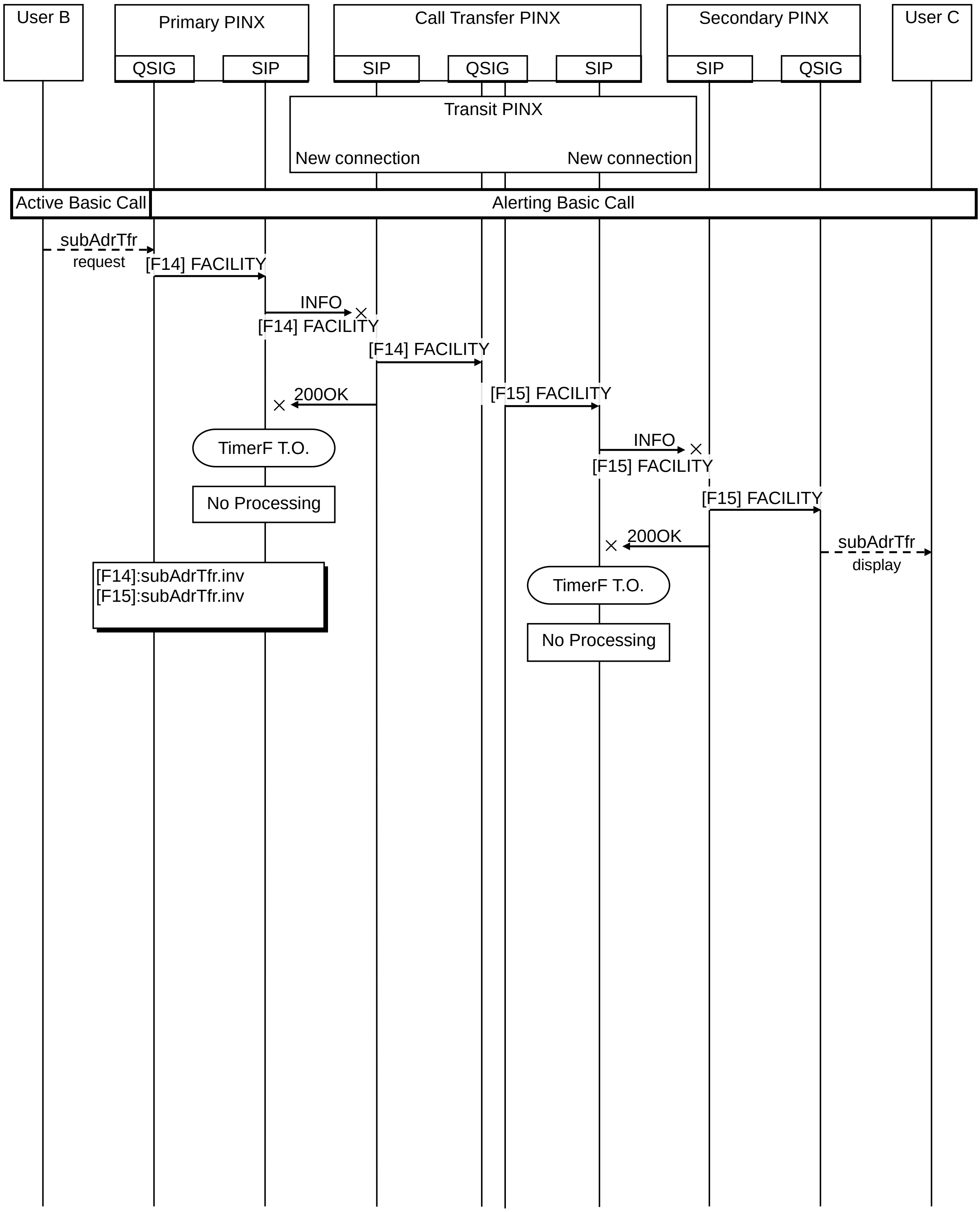
Attached Figure 3.2.4 JJ-22.06 (3/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”



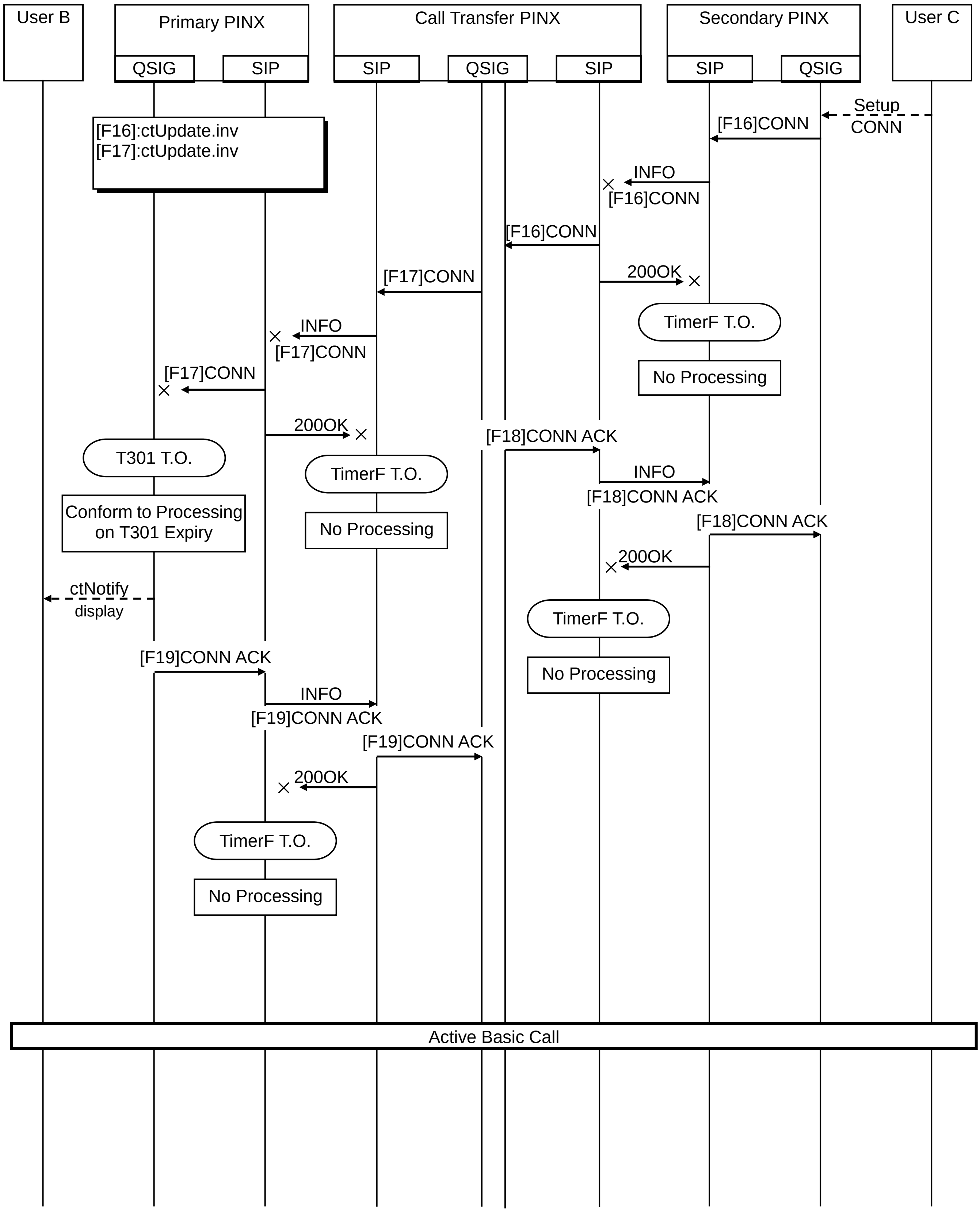
Attached Figure 3.2.4 JJ-22.06 (4/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”



Attached Figure 3.2.4 JJ-22.06 (5/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”



Attached Figure 3.2.4 JJ-22.06 (6/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”



Attached Figure 3.2.4 JJ-22.06 (7/7) Quasi-normal message sequence of rerouting-type call transfer in the case where one call is “Active” and the other is “Alerting”