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Interworking SIP and Intelligent Network (IN) Applications

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IESG Note

Abstract

Public Switched Telephone Network (PSTN) services such as 800-number routing (freephone), time-and-day routing, credit-card calling, and virtual private network (mapping a private network number into a public number) are realized by the Intelligent Network (IN). This document addresses means to support existing IN services from Session Initiation Protocol (SIP) endpoints for an IP-host-to-phone call. The call request is originated on a SIP endpoint, but the services to the call are provided by the data and procedures resident in the PSTN/IN. To provide IN services in a transparent manner to SIP endpoints, this document describes the mechanism for interworking SIP and Intelligent Network Application Part (INAP).

Table of Contents

1. Introduction

PSTN services such as 800-number routing (freephone), time-and-day routing, credit-card calling, and virtual private network (mapping a private network number into a public number) are realized by the Intelligent Network. IN is an architectural concept for the real-time execution of network services and customer applications [1]. IN is, by design, de-coupled from the call processing component of the PSTN. In this document, we describe the means to leverage this decoupling to provide IN services from SIP-based entities.

First, we will explain the basics of IN. Figure 1 shows a simplified IN architecture, in which telephone switches called Service Switching Points (SSPs) are connected via a packet network called Signaling System No. 7 (SS7) to Service Control Points (SCPs), which are general purpose computers. At certain points in a call, a switch can interrupt a call and request instructions from an SCP on how to proceed with the call. The points at which a call can be interrupted are standardized within the Basic Call State Model (BCSM) [1, 2]. The BCSM models contain two processes, one each for the originating and terminating part of a call.

When the SCP receives a request for instructions, it can reply with a single response, such as a simple number translation augmented by criteria like time of day or day of week, or, in turn, initiate a complex dialog with the switch. The situation is further complicated by the necessity to engage other specialized devices that collect digits, play recorded announcements, perform text-to-speech or speech-to-text conversions, etc. (These devices are not discussed here.) The related protocol, as well as the BCSM, is standardized by the ITU-T and known as the Intelligent Network Application Part protocol (INAP) [4]. Only the protocol, not an SCP API, has been standardized.

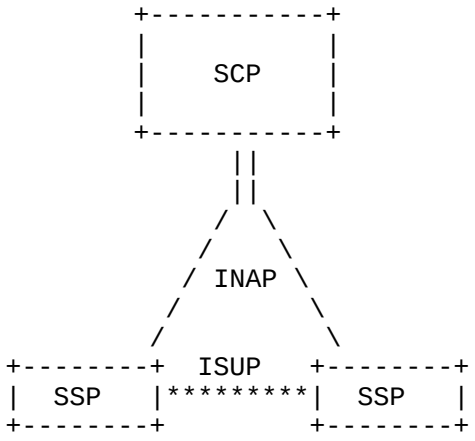


Figure 1. Simplified IN Architecture

The overall objective is to ensure that IN control of Voice over IP (VoIP) services in networks can be readily specified and implemented by adapting standards and software used in the present networks. This approach leads to services that function the same when a user connects to present or future networks, simplifies service evolution from present to future, and leads to more rapid implementation.

The rest of this document is organized as follows: Section 2 contains the architectural model of an IN aware SIP entity. Section 3 provides some issues to be taken into account when performing SIP/IN interworking (SIN). Section 4 discusses the IN service control based on the SIN approach. The technique outlined in this document focuses on the call models of IN and the SIP protocol state machine; Section 5 thus establishes a complete mapping between the two state machines that allows access to IN services from SIP endpoints. Section 6 includes call flows of IN services executing on SIP endpoints. These services are readily enabled by the technique described in this document. Finally, Section 7 covers security aspects of SIN.

List of Acronyms

B2BUA	Back-to-Back User Agent
BCSM	Basic Call State Model
CCF	Call Control Function
DP	Detection Point
DTMF	Dual Tone Multi-Frequency
IN	Intelligent Network
INAP	Intelligent Network Application Part
IP	Internet Protocol
ITU-T	International Telecommunications Union - Telecommunications Standardization Sector
O_BCSM	Originating Basic Call State Model
PIC	Point in Call
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
R-URI	Request URI
SCF	Service Control Function
SCP	Service Control Point
SIGTRAN	Signal Transport Working Group in IETF
SIN	SIP/IN Interworking
SIP	Session Initiation Protocol
SS7	Signaling System No. 7
SSF	Service Switching Function
SSP	Service Switching Point
T_BCSM	Terminating Basic Call State Model
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
VoIP	Voice over IP
VPN	Virtual Private Network

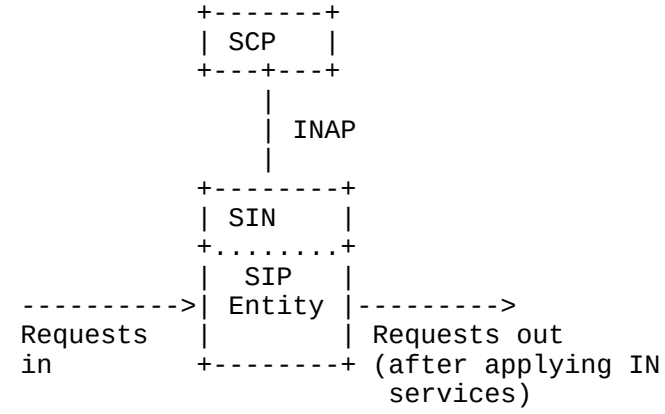
2. Access to IN-Services from a SIP Entity

The intent of this document is to provide the means to support existing IN-based applications in a SIP [3] environment. One way to gain access to IN services transparently from SIP (e.g., through the same detection points (DPs) and point-in-call (PIC) used by traditional switches) is to map the SIP protocol state machine to the IN call models [1].

From the viewpoint of IN elements such as the SCP, the request's origin from a SIP entity rather than a call processing function on a traditional switch is immaterial. Thus, it is important that the SIP entity be able to provide the same features as the traditional switch, including operating as an SSP for IN features. The SIP entity should also maintain call state and trigger queries to IN-based services, as do traditional switches.

This document does not intend to specify which SIP entity shall operate as an SSP; however, for the sake of completeness, it should be mentioned that this task should be performed by SIP entities at (or near) the core of the network rather than at the SIP end points themselves. To that extent, SIP entities such as proxy servers and Back-to-Back user agents (B2BUAs) may be employed. Generally speaking, proxy servers can be used for IN services that occur during a call setup and teardown. For IN services requiring specialized media handling (such as DTMF detection) or specialized call control (such as placing parties on hold) B2BUAs will be required.

The most expeditious manner for providing existing IN services in the IP domain is to use the deployed IN infrastructure as often as possible. In SIP, the logical point to tap into for accessing existing IN services is either the user agents or one of the proxies physically closest to the user agent (and presumably in the same administrative domain). However, SIP entities do not run an IN call model; to access IN services transparently, the trick then is to overlay the state machine of the SIP entity with an IN layer so that call acceptance and routing is performed by the native state machine and so that services are accessed through the IN layer by using an IN call model. Such an IN-enabled SIP entity, operating in synchrony with the events occurring at the SIP transaction level and interacting with the IN elements (SCP), is depicted in Figure 2:



SIN: SIP/IN Interworking layer

Figure 2. SIP Entity Accessing IN Services

Section 5 proposes this mapping between the IN layer and the SIP protocol state machine. Essentially, a SIP entity exhibiting this mapping becomes a SIN-enabled SIP entity.

This document does not propose any extensions to SIP.

Figure 3 expands the SIP entity depicted in Figure 2 and further details the architecture model involving IN and SIP interworking. Events occurring at the SIP layer will be passed to the IN layer for service application. More specifically, since IN services deal with E.164 numbers, it is reasonable to assume that a SIN-enabled SIP entity that seeks to provide services on such a number will consult the IN layer for further processing, thus acting as a SIP-based SSP. The IN layer will proceed through its BCSM states and, at appropriate points in the call, will send queries to the SCP for call disposition. Once the disposition of the call has been determined, the SIP layer is informed and processes the transaction accordingly.

Note that the single SIP entity as modeled in this figure can in fact represent several different physical instances in the network as, for example, when one SIP entity is in charge of the terminal or access network/domain, and another is in charge of the interface to the Switched Circuit Network (SCN).

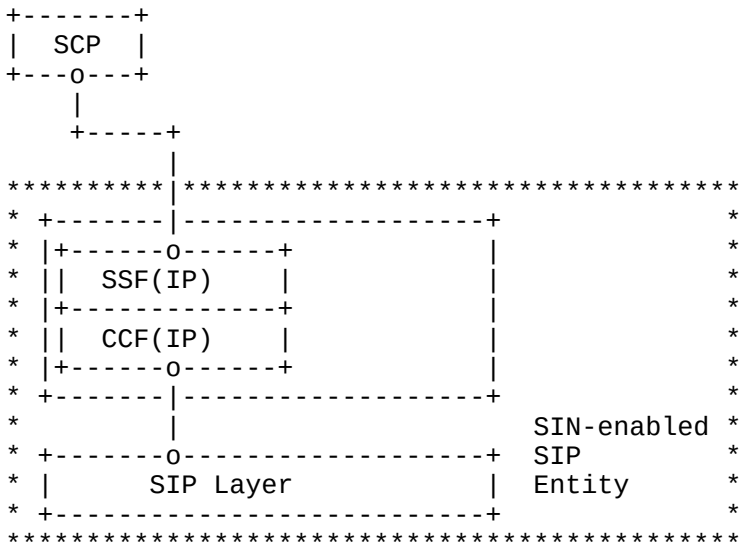


Figure 3. Functional Architecture of a SIN-Enabled SIP Entity

The following architecture entities, used in Figure 3, are defined in the Intelligent Network standards:

Service Switching Function (SSF): IN functional entity that interacts with call control functions.

Call Control Function (CCF): IN functional entity that refers to call and connection handling in the classical sense (i.e., that of an exchange).

3. Additional SIN Considerations

In working between Internet Telephony and IN-PSTN networks, the main issue is to translate between the states produced by the Internet Telephony signaling and those used in traditional IN environments. Such a translation entails attention to the considerations listed below.

3.1. The Concept of State in SIP

IN services occur within the context of a call, i.e., during call setup, call teardown, or in the middle of a call. SIP entities such as proxies, with which some of these services may be realized, typically run in transaction-stateful (or stateless) mode. In this mode, a SIP proxy that proxied the initial INVITE is not guaranteed to receive a subsequent request, such as a BYE. Fortunately, SIP has primitives to force proxies to run in a call-stateful mode; namely, the Record-Route header. This header forces the user agent client (UAC) and user agent server (UAS) to create a "route set" that consists of all intervening proxies through which subsequent requests must traverse. Thus SIP proxies must run in call-stateful mode in order to provide IN services on behalf of the UAs.

A B2BUA is another SIP element in which IN services can be realized. As a B2BUA is a true SIP UA, it maintains complete call state and is thus capable of providing IN services.

3.2. Relationship between SCP and a SIN-Enabled SIP Entity

In the architecture model proposed in this document, each SIN-enabled SIP entity is pre-configured to communicate with one logical SCP server, using whatever communication mechanism is appropriate. Different SIP servers (e.g., those in different administrative domains) may communicate with different SCP servers, so that there is no single SCP server responsible for all SIP servers.

As Figures 1 and 2 depict, the IN-portion of the SIN-enabled SIP entity will communicate with the SCP. This interface between the IN call handling layer and the SCP is not specified by this document and, indeed, can be any one of the following, depending on the interfaces supported by the SCP: INAP over IP, INAP over SIGTRAN, or INAP over SS7.

This document is only applicable when SIP-controlled Internet telephony devices seek to operate with PSTN devices. The SIP UAs using this interface would typically appear together with a media gateway. This document is *not* applicable in an all-IP network and is not needed in cases where PSTN media gateways (not speaking SIP) need to communicate with SCPs.

3.3. SIP REGISTER and IN Services

SIP REGISTER provisions a SIP Proxy or SIP Registration server. The process is similar to the provisioning of an SCP/HLR in the switched circuit network. SCPs that provide VoIP based services can leverage this information directly. However, this document neither endorses nor prohibits such an architecture and, in fact, considers it an implementation decision.

3.4. Support of Announcements and Mid-Call Signaling

Services in the IN such as credit-card calling typically play announcements and collect digits from the caller before a call is set up. Playing announcements and collecting digits require the manipulation of media streams. In SIP, proxies do not have access to the media data path. Thus, such services should be executed in a B2BUA.

Although the SIP specification [3] allows for end points to be put on hold during a call or for a change of media streams to take place, it does not have any primitives to transport other than mid-call control information. This may include transporting DTMF digits, for example. Extensions to SIP, such as the INFO method [5] or the SIP event notification extension [6], can be considered for services requiring mid-call signaling. Alternatively, DTMF can be transported in RTP itself [7].

4. The SIN Architecture

4.1. Definitions

The SIP architecture has the following functional elements defined in [3]:

- User agent client (UAC): The SIP functional entity that initiates a request.

- User agent server (UAS): The SIP functional entity that terminates a request by sending 0 or more provisional SIP responses and one final SIP response.

- Proxy server: An intermediary SIP entity that can act as both a UAS and a UAC. Acting as a UAS, it accepts requests from UACs, rewrites the Request-URI (R-URI), and, acting as a UAC, proxies the request to a downstream UAS. Proxies may retain significant call control state by inserting themselves in future SIP transactions beyond the initial INVITE.

- Redirect server: An intermediary SIP entity that redirects callers to alternate locations, after possibly consulting a location server to determine the exact location of the callee (as specified in the R-URI).

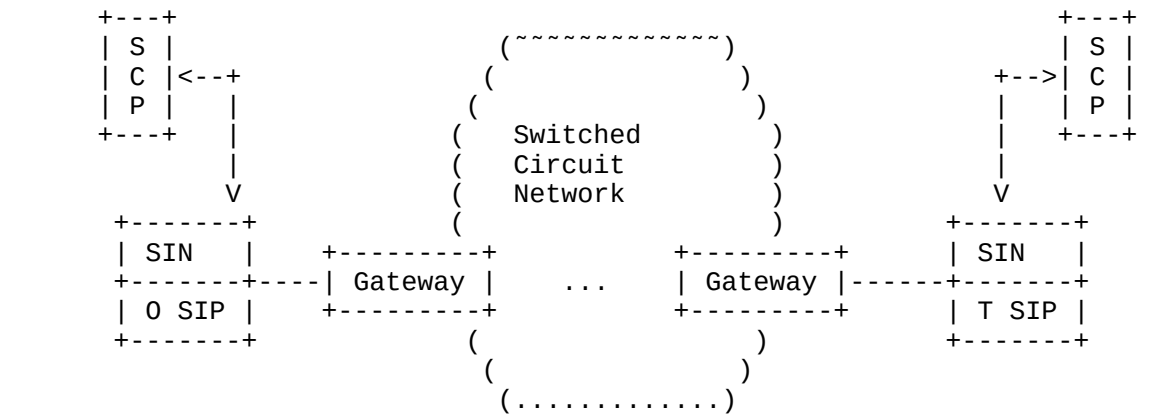
- Registrar: A SIP entity that accepts SIP REGISTER requests and maintains a binding from a high-level URL to the exact location for a user. This information is saved in some data-store that is also accessible to a SIP Proxy and a SIP Redirect server. A Registrar is usually co-located with a SIP Proxy or a SIP Redirect server.

- Outbound proxy: A SIP proxy located near the originator of requests. It receives all outgoing requests from a particular UAC, including those requests whose R-URIs identify a host other than the outbound proxy. The outbound proxy sends these requests, after any local processing, to the address indicated in the R-URI.

- Back-to-Back UA (B2BUA): A SIP entity that receives a request and processes it as a UAS. It also acts as a UAC and generates requests to determine how the incoming request is to be answered. A B2BUA maintains complete dialog state and must participate in all requests sent within the dialog.

4.2. IN Service Control Based on the SIN Approach

Figure 4 depicts the possibility of IN service control based on the SIN approach. On both the originating and terminating ends, a SIN-capable SIP entity is assumed (it can be a proxy or a B2BUA). The "O SIP" entity is required for outgoing calls that require support for existing IN services. Likewise, on the callee’s side (or terminating side), an equally configured entity ("T SIP") will be required to provide terminating side services. Note that the "O SIP" and "T SIP" entities correspond, respectively, to the IN O_BCSM and T_BCSM halves of the IN call model.



O SIP: Originating SIP entity
T SIP: Terminating SIP entity

Figure 4. Overall SIN Architecture

5. Mapping of the SIP State Machine to the IN State Model

This section establishes the mapping of the SIP protocol state machine to the IN generic basic call state model (BCSM) [2], independent of any capability sets [8, 9]. The BCSM is divided into two halves: an originating call model (O_BCSM) and a terminating call model (T_BCSM). There are a total of 19 PICs and 35 DPs between both the halves (11 PICs and 21 DPs for O_BCSM; 8 PICs and 14 DPs for T_BCSM) [1]. The SSPs, SCPs, and other IN elements track a call's progress in terms of the basic call model. The basic call model provides a common context for communication about a call.

O_BCSM has 11 PICs:

O_NULL: Starting state; call does not exist yet.
AUTH_ORIG_ATTEMPT: Switch detects a call setup request.
COLLECT_INFO: Switch collects the dial string from the calling party.
ANALYZE_INFO: Complete dial string is translated into a routing address.
SELECT_ROUTE: Physical route is selected, based on the routing address.
AUTH_CALL_SETUP: Switch ensures the calling party is authorized to place the call.
CALL_SENT: Control of call sent to terminating side.
O_ALERTING: Switch waits for the called party to answer.
O_ACTIVE: Connection established; communications ensue.
O_DISCONNECT: Connection torn down.
O_EXCEPTION: Switch detects an exceptional condition.

T_BCSM has 8 PICS:

T_NULL: Starting state; call does not exist yet.
AUTH_TERM_ATT: Switch verifies whether the call can be sent to terminating party.
SELECT_FACILITY: Switch picks a terminating resource to send the call on.
PRESENT_CALL: Call is being presented to the called party.
T_ALERTING: Switch alerts the called party, e.g., by ringing the line.
T_ACTIVE: Connection established; communications ensue.
T_DISCONNECT: Connection torn down.
T_EXCEPTION: Switch detects an exceptional condition.

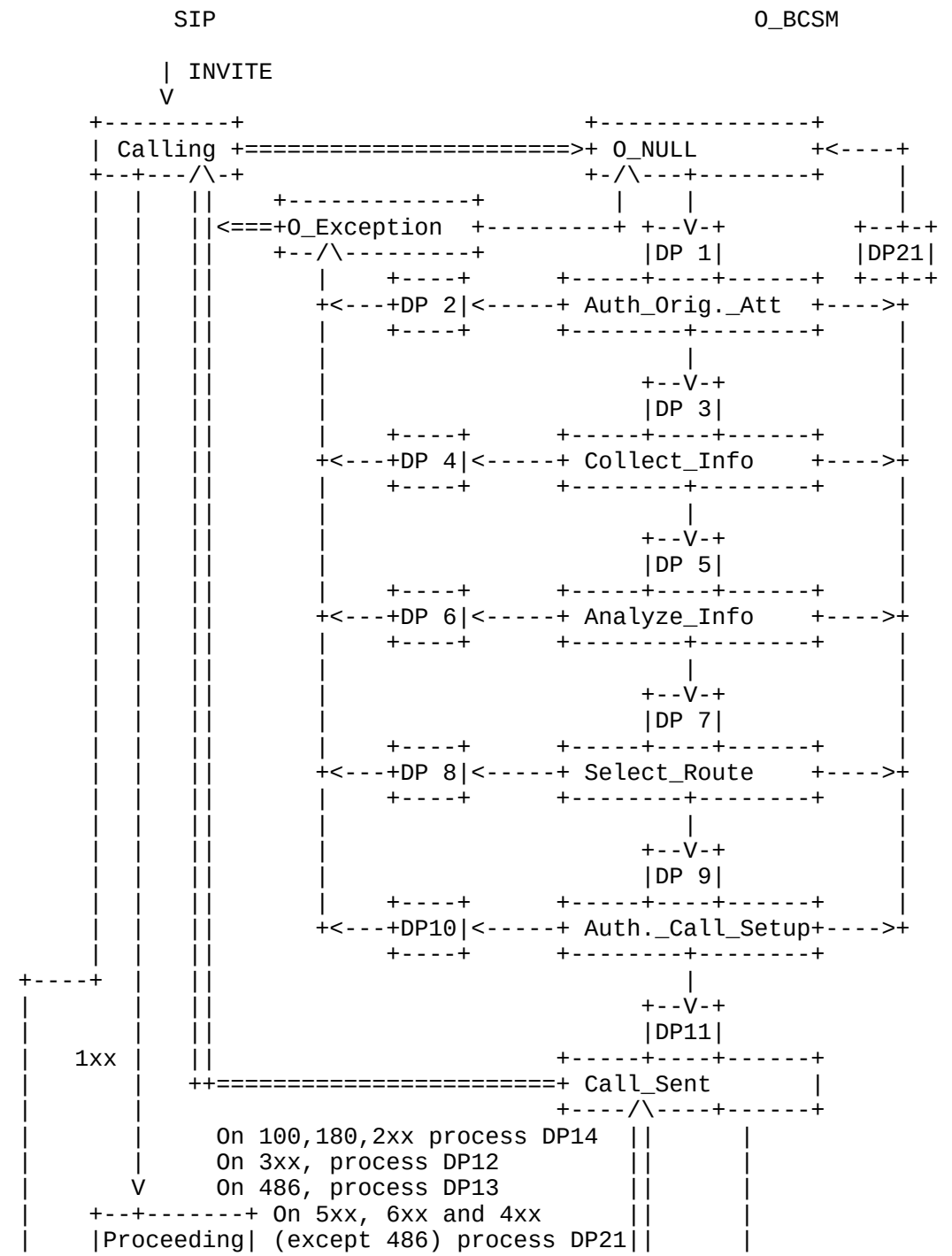
The state machine for O_BCSM and T_BCSM is provided in [1] on pages 98 and 103, respectively. This state machine will be used for subsequent discussion when the IN call states are mapped into SIP.

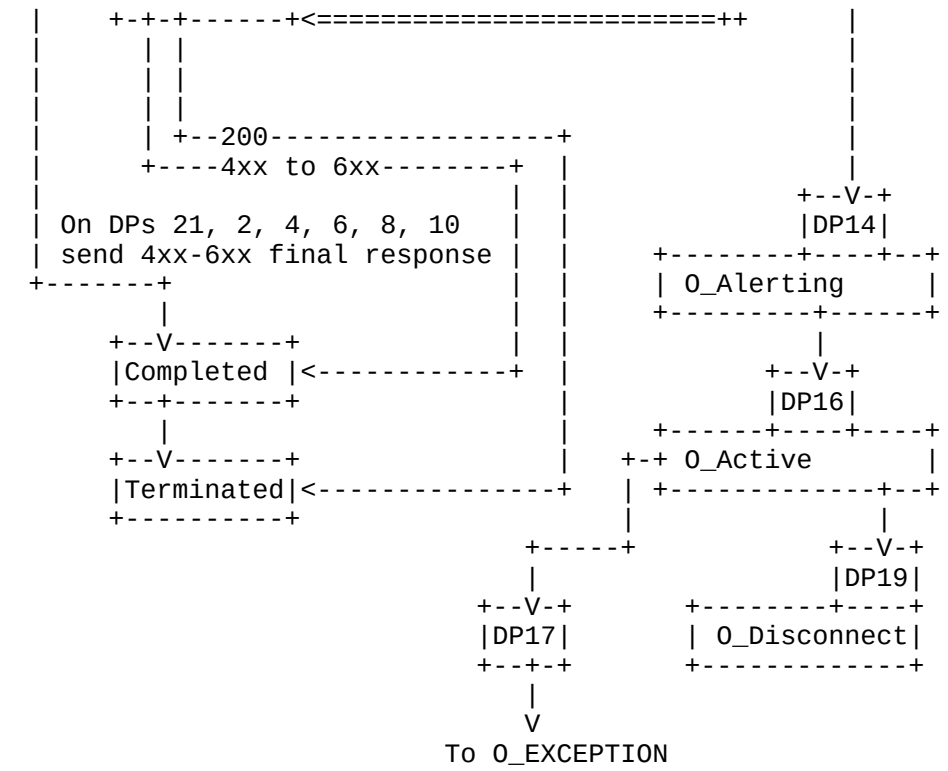
The next two sections contain the mapping of the SIP protocol state machine to the IN BCSMs. Explaining all PICs and DPs in an IN call model is beyond the scope of this document. It is assumed that the reader has some familiarity with the PICs and DPs of the IN call model. More information can be found in [1]. For a quick reference, Appendix A contains a mapping of the DPs to the SIP response codes as discussed in the next two sections.

5.1. Mapping SIP Protocol State Machine to O_BCSM

The 11 PICs of O_BCSM come into play when a call request (SIP INVITE message) arrives from an upstream SIP client to an originating SIN-enabled SIP entity running the IN call model. This entity will create an O_BCSM object and initialize it in the O_NULL PIC. The next seven IN PICs -- O_NULL, AUTH_ORIG_ATT, COLLECT_INFO, ANALYZE_INFO, SELECT_ROUTE, AUTH_CALL_SETUP, and CALL_SENT -- can all be mapped to the SIP "Calling" state.

Figure 5 provides a visual map from the SIP protocol state machine to the originating half of the IN call model. Note that control of the call shuttles between the SIP protocol machine and the IN O_BCSM call model while it is being serviced.


SIP
O_BCSM
INVITE
Calling
O_NULL
O_Exception
DP 1
DP21
DP 2
Auth_Orig._Att
DP 3
DP 4
Collect_Info
DP 5
DP 6
Analyze_Info
DP 7
DP 8
Select_Route
DP 9
DP10
Auth._Call_Setup
DP11
Call_Sent
1xx
On 100,180,2xx process DP14
On 3xx, process DP12
On 486, process DP13
On 5xx, 6xx and 4xx
(except 486) process DP21
Proceeding



Legend:

| Communication between
| states in the same
V protocol

======> Communication between IN Layer and SIP Protocol
State machine to transfer call state

Figure 5. Mapping from SIP to O_BCSM

The SIP "Calling" protocol state has enough functionality to absorb the seven PICs as described below:

O_NULL: This PIC is basically a fall through state to the next PIC, AUTHORIZE_ORIGINATION_ATTEMPT.

AUTHORIZE_ORIGINATION_ATTEMPT: In this PIC, the IN layer has detected that someone wishes to make a call. Under some circumstances (e.g., if the user is not allowed to make calls during certain hours), such a call cannot be placed. SIP can authorize the calling party by using a set of policy directives

configured by the SIP administrator. If the called party is authorized to place the call, the IN layer is instructed to enter the next PIC, COLLECT_INFO through DP 3 (Origination_Attempt_Authorized). If for some reason the call cannot be authorized, DP 2 (Origination_Denied) is processed, and control transfers to the SIP state machine. The SIP state machine must format and send a non-2xx final response (possibly 403) to the upstream entity.

COLLECT_INFO: This PIC is responsible for collecting a dial string from the calling party and verifying the format of the string. If overlap dialing is being used, this PIC can invoke DP 4 (Collect_Timeout) and transfer control to the SIP state machine, which will format and send a non-2xx final response (possibly a 484). If the dial string is valid, DP 5 (Collected_Info) is processed, and the IN layer is instructed to enter the next PIC, ANALYZE_INFO.

ANALYZE_INFO: This PIC is responsible for translating the dial string to a routing number. Many IN services, such as freephone, LNP (Local Number Portability), and OCS (Originating Call Screening) occur during this PIC. The IN layer can use the R-URI of the SIP INVITE request for analysis. If the analysis succeeds, the IN layer is instructed to enter the next PIC, SELECT_ROUTE. If the analysis fails, DP 6 (Invalid_Info) is processed, and the control transfers to the SIP state machine, which will generate a non-2xx final response (possibly 400, 401, 403, 404, 405, 406, 410, 414, 415, 416, 485, or 488) and send it to the upstream entity.

SELECT_ROUTE: In the circuit-switched network, the actual physical route has to be selected at this point. The SIP analogue would be to determine the next hop SIP server. This could be chosen by a variety of means. For instance, if the Request URI in the incoming INVITE request is an E.164 number, the SIP entity can use a protocol like TRIP [10] to find the best gateway to egress the request onto the PSTN. If a successful route is selected, the IN call model moves to PIC AUTH_CALL_SETUP via DP 9 (Route_Selected). Otherwise, the control transfers to the SIP state machine via DP 8 (Route_Select_Failure), which will generate a non-2xx final response (possibly 488) and send it to the upstream entity.

AUTH_CALL_SETUP: Certain service features restrict the type of call that may originate on a given line or trunk. This PIC is the point at which relevant restrictions are examined. If no such restrictions are encountered, the IN call model moves to PIC CALL_SENT via DP 11 (Origination_Authorized). If a restriction is encountered that prohibits further processing of the call, DP 10

(Authorization_Failure) is processed, and control is transferred to the SIP state machine, which will generate a non-2xx final response (possibly 404, 488, or 502). Otherwise, DP 11 (Origination_Authorized) is processed, and the IN layer is instructed to enter the next PIC, CALL_SENT.

CALL_SENT: At this point, the request needs to be sent to the downstream entity. The IN layer waits for a signal confirming either that the call has been presented to the called party or that a called party cannot be reached for a particular reason. The control is transferred to the SIP state machine. The SIP state machine should now send the call to the next downstream server determined in PIC SELECT_ROUTE. The IN call model now blocks until unblocked by the SIP state machine.

If the above seven PICs have been successfully negotiated, the SIN-enabled SIP entity now sends the SIP INVITE message to the next hop server. Further processing now depends on the provisional responses (if any) and the final response received by the SIP protocol state machine. The core SIP specification does not guarantee the delivery of 1xx responses; thus special processing is needed at the IN layer to transition to the next PIC (O_ALERTING) from the CALL_SENT PIC. The special processing needed for responses while the SIP state machine is in the "Proceeding" state and the IN layer is in the "CALL_SENT" state is described next.

A 100 response received at the SIP state machine elicits no special behavior in the IN layer.

A 180 response received at the SIP entity enables the processing of DP 14 (O_Term_Seized), however, a state transition to O_ALERTING is not undertaken yet. Instead, the IN layer is instructed to remain in the CALL_SENT PIC until a final response is received.

A 2xx response received at the SIP entity enables the processing of DP 14 (O_Term_Seized), and the immediate transition to the next state, O_ALERTING (processing in O_ALERTING is described later).

A 3xx response received at the SIP entity enables the processing of DP 12 (Route_Failure). The IN call model from this point goes back to the SELECT_ROUTE PIC to select a new route for the contacts in the 3xx final response (not shown in Figure 5 for brevity).

A 486 (Busy Here) response received at the SIP entity enables the processing of DP 13 (O_Called_Party_Busy) and resources for the call are released at the IN call model.

If the SIN-enabled SIP entity gets a 4xx (except 486), 5xx, or 6xx final response, DP 21 (O_Calling_Party_Disconnect & O_Abandon) is processed and control passes to the SIP state machine. Since a call was not successfully established, both the IN layer and the SIP state machine can release resources for the call.

O_ALERTING - This PIC will be entered as a result of receiving a 200-class response. Since a 200-class response to an INVITE indicates acceptance, this PIC is mostly a fall through to the next PIC, O_ACTIVE via DP 16 (O_Answer).

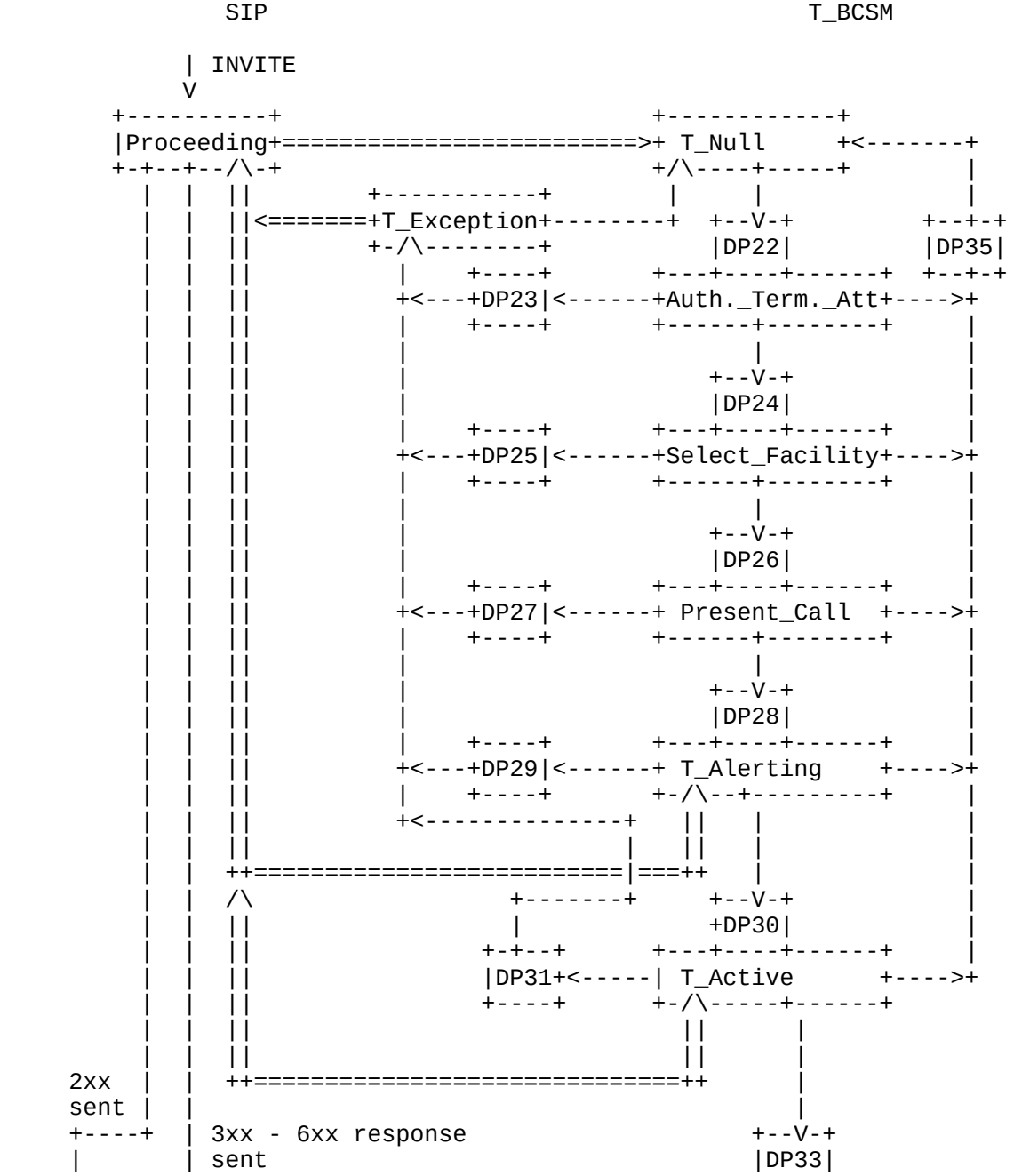
O_ACTIVE - At this point, the call is active. Once in this state, the call may get disconnected only when one of the following three events occur: (1) the network connection fails, (2) the called party disconnects the call, or (3) the calling party disconnects the call. If event (1) occurs, DP 17 (O_Connection_Failure) is processed and call control is transferred to the SIP protocol state machine. Since the network failed, there is not much sense in attempting to send a BYE request; thus, both the SIP protocol state machine and the IN call layer should release all resources associated with the call and initialize themselves to the null state. Event (2) results in the processing of DP 19 (O_DISCONNECT) and a move to the last PIC, O_DISCONNECT. Event (3) occurs if the calling party deliberately terminated the call. In this case, DP 21 (O_Abandon & O_Calling_Party_Disconnect) will be processed, and control will be passed to the SIP protocol state machine. The SIP protocol state machine must send a BYE request and wait for a final response. The IN layer releases all of its resources and initializes itself to the null state.

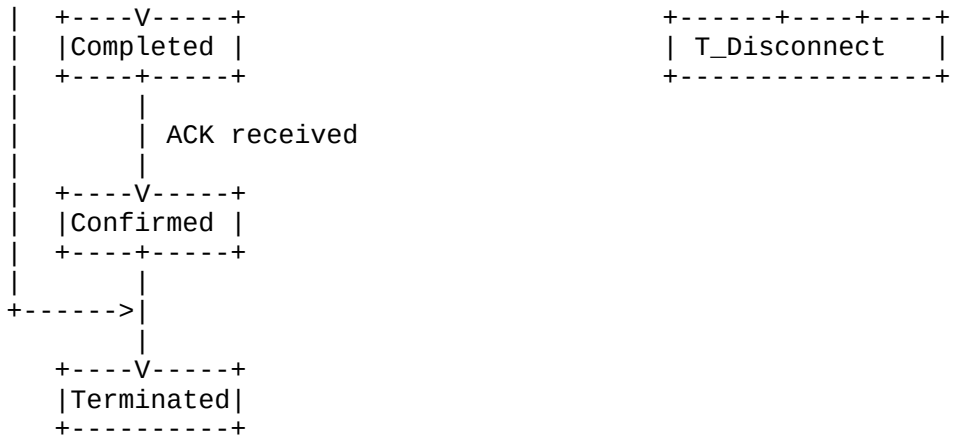
O_DISCONNECT: When the SIP entity receives a BYE request, the IN layer is instructed to move to the last PIC, O_DISCONNECT via DP 19. A final response for the BYE is generated and transmitted by the SIP entity, and the call resources are freed by both the SIP protocol state machine and the IN layer.

5.2. Mapping SIP Protocol State Machine to T_BCSM

The T_BCSM object is created when a SIP INVITE message makes its way to the terminating SIN-enabled SIP entity. This entity creates the T_BCSM object and initializes it to the T_NULL PIC.

Figure 6 provides a visual map from the SIP protocol state machine to the terminating half of the IN call model:





Legend:

| Communication between
| states in the same
V protocol
======> Communication between IN call model and SIP
protocol state machine to transfer call state

Figure 6. Mapping from SIP to T_BCSM

The SIP "Proceeding" state has enough functionality to absorb the first five PICS -- T_Null, Authorize_Termination_Attempt, Select_Facility, Present_Call, T_Alerting -- as described below:

T_NULL: At this PIC, the terminating end creates the call at the IN layer. The incoming call results in the processing of DP 22, Termination_Attempt, and a transition to the next PIC, AUTHORIZE_TERMINATION_ATTEMPT, takes place.

AUTHORIZE_TERMINATION_ATTEMPT: At this PIC, it is ascertained that the called party wishes to receive the call and that the facilities of the called party are compatible with those of the calling party. If any of these conditions is not met, DP 23 (Termination_Denied) is invoked, and the call control is transferred to the SIP protocol state machine. The SIP protocol state machine can format and send a non-2xx final response (possibly 403, 405, 415, or 480). If the conditions of the PIC are met, processing of DP 24 (Termination_Authorized) is invoked, and a transition to the next PIC, SELECT_FACILITY, takes place.

SELECT_FACILITY: In circuit switched networks, this PIC is intended to select a line or trunk to reach the called party. As lines or trunks are not applicable in an IP network, a SIN-enabled SIP entity can use this PIC to interface with a PSTN gateway and select a line/trunk to route the call. If the called party is busy, or if a line/trunk cannot be seized, the processing of DP 25 (T_Called_Party_Busy) is invoked, and the call goes to the SIP protocol state machine. The SIP protocol state machine must format and send a non-2xx final response (possibly 486 or 600). If a line/trunk was successfully seized, the processing of DP 26 (Terminating_Resource_Available) is invoked, and a transition to the next PIC, PRESENT_CALL, takes place.

PRESENT_CALL: At this point, the call is being presented (via the ISUP ACM message, or Q.931 Alerting message, or simply by ringing a POTS phone). If there was an error presenting the call, the processing of DP 27 (Presentation_Failure) is invoked, and the call control is transferred to the SIP protocol state machine, which must format and send a non-2xx final response (possibly 480). If the call was successfully presented, the processing of DP 28 (T_Term_Seized) is invoked, and a transition to the next PIC, T_ALERTING, takes place.

T_ALERTING: At this point, the called party is being "alerted". Control now passes momentarily to the SIP protocol state machine so that it can generate and send a "180 Ringing" response to its peer. Furthermore, since network resources have been allocated for the call, timers are set to prevent indefinite holding of such resources. The expiration of the relevant timers results in the processing of DP 29 (T_No_Answer), and the call control is transferred to the SIP protocol state machine, which must format and send a non-2xx final response (possibly 408). If the called party answers, then DP 30 (T_Answer) is processed, followed by a transition to the next PIC, T_ACTIVE.

After the above five PICs have been negotiated, the rest are mapped as follows:

T_ACTIVE: The call is now active. Once this state is reached, the call may become inactive under one of the following three conditions: (1) The network fails the connection, (2) the called party disconnects the call, or (3) the calling party disconnects the call. Event (1) results in the processing of DP 31 (T_Connection_Failure), and call control is transferred to the SIP protocol state machine. Since the network failed, there is little sense in attempting to send a BYE request; thus, both the SIP protocol state machine and the IN call layer should release all resources associated with the call and initialize themselves to

the null state. Event (2) results in the processing of DP 33 (T_Disconnect) and a transition to the next PIC, T_DISCONNECT. Event (3) occurs at the receipt of a BYE request at the SIP protocol state machine (not shown in Figure 6). Resources for the call should be deallocated, and the SIP protocol state machine must send a 200 OK for the BYE request (not shown in Figure 6).

T_DISCONNECT: In this PIC, the disconnect treatment associated with the called party's having disconnected the call is performed at the IN layer. The SIP protocol state machine sends out a BYE and awaits a final response for the BYE (not shown in Figure 6).

6. Examples of Call Flows

Two examples are provided here to show how SIP protocol state machine and the IN call model work synchronously with each other.

In the first example, a SIP UAC originates a call request destined to an 800 freephone number:

```
INVITE sip:18005551212@example.com SIP/2.0
From: sip:16305551212@example.net;tag=991-7as-66ff
To: sip:18005551212@example.com
Via: SIP/2.0/UDP stn1.example.net
Call-ID: 67188121@example.net
CSeq: 1 INVITE
```

The request makes its way to the originating SIP network server running an IN call model. The SIP network server hands, at the very least, the To: field and the From: field to the IN layer for freephone number translation. The IN layer proceeds through its PICs and at the ANALYSE_INFO PIC consults the SCP for freephone translation. The translated number is returned to the SIP network server, which forwards the message to the next hop SIP proxy, with the freephone number replaced by the translated number:

```
INVITE sip:18475551212@example.com SIP/2.0
From: sip:16305551212@example.net;tag=991-7as-66ff
To: sip:18005551212@example.com
Via: SIP/2.0/UDP ext-stn2.example.net
Via: SIP/2.0/UDP stn1.example.net
Call-ID: 67188121@example.net
CSeq: 1 INVITE
```

In the next example, a SIP UAC originates a call request destined to a 900 number:

```
INVITE sip:19005551212@example.com SIP/2.0
From: sip:16305551212@example.net;tag=991-7as-66dd
To: sip:19005551212@example.com
Via: SIP/2.0/UDP stn1.example.net
Call-ID: 88112@example.net
CSeq: 1 INVITE
```

The request makes its way to the originating SIP network server running an IN call model. The SIP network server hands, at the very least, the To: field and the From: field to the IN layer for 900 number translation. The IN layer proceeds through its PICs and at the ANALYSE_INFO PIC consults the SCP for the translation. During the translation, the SCP detects that the originating party is not allowed to make 900 calls. It passes this information to the originating SIP network server, which informs the SIP UAC by using a SIP "403 Forbidden" response status code:

```
SIP/2.0 403 Forbidden
From: sip:16305551212@example.net;tag=991-7as-66dd
To: sip:19005551212@example.com;tag=78K-909II
Via: SIP/2.0/UDP stn1.example.net
Call-ID: 88112@example.net
CSeq: 1 INVITE
```

7. Security Considerations

Security considerations for SIN services cover both networks being used, namely, the PSTN and the Internet. SIN uses the security measures in place for both the networks. With reference to Figure 2, the INAP messages between the SCP and the SIN-enabled SIP entity must be secured by the signaling transport used between the SCP and the SIN-enabled entity. Likewise, the requests coming into the SIN-enabled SIP entity must first be authenticated and, if need be, encrypted as well, using the means and procedures defined in [3] for SIP requests.

8. References

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Appendix A: Mapping of 4xx-6xx Responses in SIP to IN Detections Points

The mapping of error codes 4xx-6xx responses in SIP to the possible Detection Points in PIC Originating and Terminating Call Handling is indicated in the table below. The reason phrase in the 4xx-6xx response is reproduced from [3].

SIP response code	DP mapping to IN
200 OK	DP 14
3xx	DP 12
403 Forbidden	DP 2, DP 21
484 Address Incomplete	DP 4, DP 21
400 Bad Request	DP 6, DP 21
401 Unauthorized	DP 6, DP 21
403 Forbidden	DP 6, DP 21, DP 23
404 Not Found	DP 6, DP 21
405 Method Not Allowed	DP 6, DP 21, DP 23
406 Not Acceptable	DP 6, DP 21
408 Request Timeout	DP 29
410 Gone	DP 6, DP 21
414 Request-URI Too Long	DP 6, DP 21
415 Unsupported Media Type	DP 6, DP 21, DP 23
416 Unsupported URI Scheme	DP 6, DP 21
480 Temporarily Unavailable	DP 23, DP 27
485 Ambiguous	DP 6, DP 21
486 Busy Here	DP 13, DP 21, DP 25
488 Not Acceptable Here	DP 6, DP 21

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