

PJSIP

# Developer's Guide

## **ABOUT PJSIP**

---

PJSIP is small-footprint and high-performance SIP stack written in C.  
PJSIP is distributed under dual licensing schemes: GPL and commercial license.  
Please visit <http://www.pjproject.net> for more details.

## **ABOUT THIS DOCUMENT**

---

### **Copyright ©2005-2006 Benny Prijono**

This is a free document distributed under GNU Free Documentation License version 1.2. Everyone is permitted to copy and distribute verbatim copies of this document, but changing it is not allowed.

## **DOCUMENT REVISION HISTORY**

---

<b>Version</b>	<b>Date</b>	<b>Author</b>	<b>Description</b>
0.5	19 Dec 2005	bennylp	Initial revision

# Table of Contents

<b>TABLE OF CONTENTS.....</b>	<b>3</b>
<b>TABLE OF FIGURES.....</b>	<b>6</b>
<b>TABLE OF CODES.....</b>	<b>6</b>
<b>CHAPTER 1: GENERAL DESIGN.....</b>	<b>8</b>
1.1 ARCHITECTURE.....	8
1.1.1 Communication Diagram.....	8
1.1.2 Class Diagram.....	8
1.2 MODULE.....	9
1.2.1 Module Declaration.....	9
1.2.2 Module Priorities.....	10
1.2.3 Message Processing by Modules.....	10
1.2.4 Transaction User and State Callback.....	11
1.2.5 Module Specific Data.....	11
1.2.6 Callback Summary.....	12
1.2.7 Sample Callback Requirements for Applications.....	12
1.2.8 Sample Callback Diagrams.....	13
Incoming Message Outside Transaction and Outside Dialog.....	13
Incoming Message Inside Transaction.....	13
Incoming Message Inside Dialog but Outside Transaction.....	14
1.3 MODULE MANAGEMENT.....	14
1.3.1 Module Management API.....	14
<b>CHAPTER 2: MESSAGE ELEMENTS.....</b>	<b>15</b>
2.1 UNIFORM RESOURCE INDICATOR (URI).....	15
2.1.1 URI "Class Diagram".....	15
2.1.2 URI Context.....	15
2.1.3 Base URI.....	16
2.1.4 SIP and SIPS URI.....	17
2.1.5 Tel URI.....	17
2.1.6 Name Address.....	18
2.1.7 Sample URI Manipulation Program.....	18
2.2 SIP METHODS.....	20
2.2.1 SIP Method Representation ( <i>pjsip_method</i> ).....	20
2.2.2 SIP Method API.....	20
2.3 HEADER FIELDS.....	21
2.3.1 Header "Class Diagram".....	21
2.3.2 Header Structure.....	22
2.3.3 Common Header Functions.....	22
2.3.4 Supported Header Fields.....	23
2.3.5 Header Array Elements.....	23
2.4 MESSAGE BODY ( <i>PJSIP_MSG_BODY</i> ).....	24
2.5 MESSAGE ( <i>PJSIP_MSG</i> ).....	25
2.6 SIP STATUS CODES.....	26
2.7 NON-STANDARD PARAMETER ELEMENTS.....	27
2.7.1 Data Structure Representation ( <i>pjsip_param</i> ).....	28
2.7.2 Non-Standard Parameter Manipulation.....	28
2.8 ESCAPEMENT RULES.....	28
<b>CHAPTER 3: PARSER.....</b>	<b>30</b>
3.1 FEATURES.....	30
3.2 FUNCTIONS.....	31
3.2.1 Message Parsing.....	31
3.2.2 URI Parsing.....	31
3.2.3 Header Parsing.....	31
3.3 EXTENDING PARSER.....	32

<b>CHAPTER 4: MESSAGE BUFFERS.....</b>	<b>33</b>
4.1 RECEIVE DATA BUFFER.....	33
4.1.1 Receive Data Buffer Structure.....	33
4.2 TRANSMIT DATA BUFFER (PJSIP_TX_DATA).....	34
<b>CHAPTER 5: TRANSPORTS.....</b>	<b>35</b>
5.1 TRANSPORT CLASS DIAGRAM.....	35
5.2 USING TRANSPORTS.....	35
5.2.1 Function Reference.....	35
5.3 EXTENDING TRANSPORTS.....	36
5.4 INITIALIZING TRANSPORTS.....	36
5.4.1 UDP Transport Initialization.....	36
5.4.2 TCP Transport Initialization.....	37
5.4.3 TLS Transport Initialization.....	37
5.4.4 SCTP Transport Initialization.....	37
<b>CHAPTER 6: STATELESS OPERATIONS.....</b>	<b>38</b>
6.1 SENDING STATELESS RESPONSE.....	38
6.1.1 Base Functions.....	38
6.1.2 Composite Functions.....	38
6.1.3 Usage Examples.....	39
Sending Account Not Found Response Statelessly.....	39
Handling Authentication Failures Statelessly.....	39
Stateless Redirection.....	39
6.2 SENDING STATELESS REQUEST.....	40
6.2.1 Related Functions.....	40
6.2.2 Usage Examples.....	40
Sending Request Statelessly.....	40
6.3 STATELESS PROXY FORWARDING.....	41
6.3.1 Base Functions.....	41
6.3.2 Usage Examples.....	42
Stateless Forwarding.....	42
<b>CHAPTER 7: TRANSACTIONS.....</b>	<b>43</b>
7.1 DESIGN.....	43
7.1.1 Introduction.....	43
7.1.2 Timers and Retransmissions.....	43
7.1.3 INVITE Final Response and ACK Request.....	43
Failed INVITE Request.....	43
Successful INVITE Request.....	44
7.1.4 Incoming ACK Request.....	44
7.1.5 Server Resolution and Transports.....	44
TCP Connection Closure.....	45
7.1.6 Via Header.....	45
Branch Parameter.....	45
Via Sent-By.....	45
7.2 REFERENCE.....	45
7.2.1 Base Functions.....	45
7.2.2 Composite Functions.....	46
7.3 SENDING STATEFULL RESPONSES.....	46
7.3.1 Usage Examples.....	46
Sending Response Statefully (The Hard Way).....	46
Sending Response Statefully (The Easy Way).....	47
7.4 SENDING STATEFULL REQUEST.....	47
7.4.1 Usage Examples.....	47
Sending Request with Completion Callback.....	47
Sending Request with Transaction.....	48
7.5 STATEFULL PROXY FORWARDING.....	48
7.5.1 Usage Examples.....	48
Statefull Forwarding.....	48

<b>CHAPTER 8: AUTHENTICATION FRAMEWORK.....</b>	<b>50</b>
8.1 CLIENT AUTHENTICATION FRAMEWORK.....	50
8.1.1 Client Authentication Framework Reference.....	50
Data Structure Reference.....	50
Function Reference.....	51
8.1.2 Examples.....	51
Client Transaction Authentication.....	51
8.2 SERVER AUTHORIZATION FRAMEWORK.....	52
8.2.1 Server Authorization Reference.....	52
Data Types Reference.....	52
Functions Reference.....	53
8.3 EXTENDING AUTHENTICATION FRAMEWORK.....	53
<b>CHAPTER 9: BASIC USER AGENT LAYER (UA).....</b>	<b>54</b>
9.1 BASIC DIALOG CONCEPT.....	54
9.1.1 Dialog Sessions.....	54
9.1.2 Dialog Usages.....	54
9.1.3 Class Diagram.....	55
9.1.4 Forking.....	56
Handling Forking Condition.....	56
Creating Forked Dialog.....	56
Using Timer to Handle Failed Forked Dialog.....	57
9.1.5 CSeq Sequencing.....	57
9.1.6 Authentication.....	58
9.1.7 Stateless Operations.....	58
9.2 BASIC UA API REFERENCE.....	58
9.2.1 Dialog Creation API.....	58
9.2.2 Dialog Session Management API.....	59
9.2.3 Dialog Usages API.....	59
9.2.4 Dialog Request and Response API.....	59
9.2.5 Dialog Auxiliary API.....	60
9.3 EXAMPLES.....	61
9.3.1 Incoming Invite Dialog.....	61
Creating Initial Invite Dialog.....	61
Answering Dialog.....	62
Processing CANCEL Request.....	62
Processing ACK Request.....	63
9.3.2 Outgoing Invite Dialog.....	63
Creating Initial Dialog.....	63
Receiving Response.....	64
Sending ACK.....	64
<b>CHAPTER 10: SDP OFFER/ANSWER FRAMEWORK.....</b>	<b>66</b>
10.1 SDP NEGOTIATOR STRUCTURE.....	66
10.2 SDP NEGOTIATOR SESSION.....	67
10.3 SDP NEGOTIATION FUNCTION.....	68
<b>CHAPTER 11: DIALOG INVITE USAGE.....</b>	<b>69</b>
11.1 INTRODUCTION.....	69
11.1.1 Invite Session State.....	69
11.1.2 Invite Usage "Class Diagram".....	70
<b>CHAPTER 12: DIALOG SUBSCRIBE USAGE.....</b>	<b>71</b>

## Table of Figures

FIGURE 1 COLLABORATION DIAGRAM.....	8
FIGURE 2 CLASS DIAGRAM.....	8
FIGURE 3 MODULE STATE DIAGRAM.....	9
FIGURE 4 CASCADE MODULE CALLBACK.....	10
FIGURE 5 CALLBACK SUMMARY.....	12
FIGURE 6 SAMPLE CALLBACK REQUIREMENTS.....	12
FIGURE 7 PROCESSING OF INCOMING MESSAGE OUTSIDE TRANSACTION/DIALOG .....	13
FIGURE 8 PROCESSING OF INCOMING MESSAGE INSIDE TRANSACTION.....	13
FIGURE 9 PROCESSING OF INCOMING MESSAGE INSIDE DIALOG BUT OUTSIDE TRANSACTION.....	14
FIGURE 10 URI "CLASS DIAGRAM".....	15
FIGURE 11 HEADER "CLASS DIAGRAM".....	21
FIGURE 12 AUTHENTICATION FRAMEWORK.....	50
FIGURE 13 CLIENT AUTHENTICATION DATA STRUCTURE.....	50
FIGURE 14 BASIC USER AGENT CLASS DIAGRAM.....	55
FIGURE 15 SDP NEGOTIATOR "CLASS DIAGRAM".....	66
FIGURE 16 SDP OFFER/ANSWER SESSION STATE DIAGRAM.....	67
FIGURE 17 INVITE SESSION STATE DIAGRAM.....	69
FIGURE 18 INVITE SESSION STATE DESCRIPTION.....	70
FIGURE 19 DIALOG INVITE USAGE "CLASS DIAGRAM".....	70

## Table of Codes

CODE 1 MODULE DECLARATION.....	9
CODE 2 MODULE PRIORITIES.....	10
CODE 3 MODULE SPECIFIC DATA.....	11
CODE 4 ACCESSING MODULE SPECIFIC DATA.....	11
CODE 5 URI CONTEXT.....	15
CODE 6 GENERIC URI DECLARATION.....	16
CODE 7 URI VIRTUAL FUNCTION TABLE.....	16
CODE 8 SIP URI DECLARATION.....	17
CODE 9 TEL URI DECLARATION.....	18
CODE 10 NAME ADDRESS DECLARATION.....	18
CODE 11 SAMPLE URI MANIPULATION PROGRAM.....	19
CODE 12 SIP METHOD DECLARATION.....	20
CODE 13 SIP METHOD ID.....	20
CODE 14 GENERIC HEADER DECLARATION.....	22

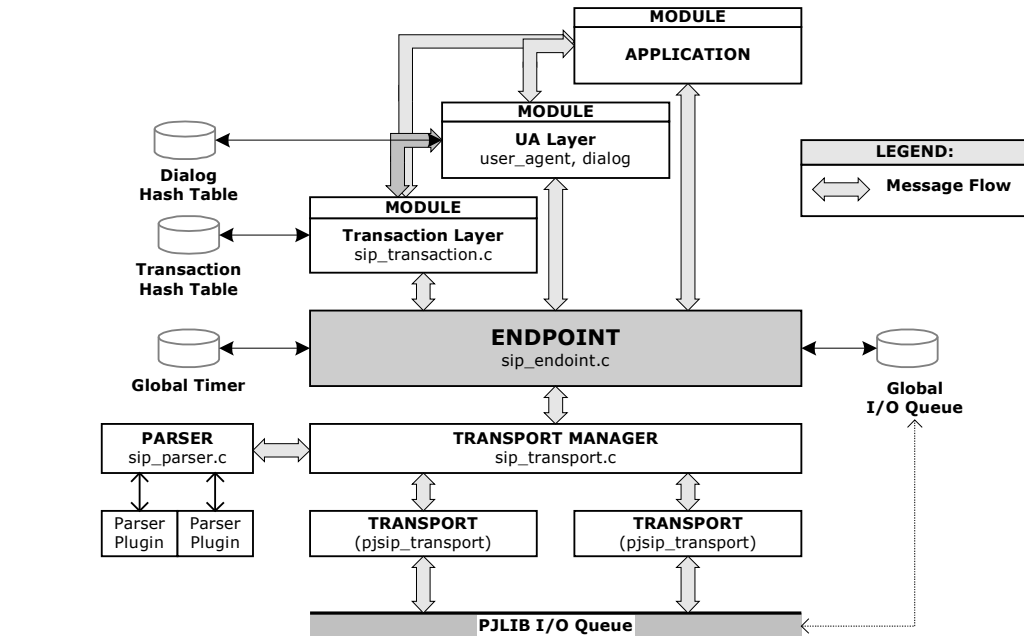
<b>CODE 15 GENERIC HEADER DECLARATION.....</b>	<b>22</b>
<b>CODE 16 HEADER VIRTUAL FUNCTION TABLE.....</b>	<b>22</b>
<b>CODE 17 MESSAGE BODY DECLARATION.....</b>	<b>25</b>
<b>CODE 18 SIP MESSAGE DECLARATION.....</b>	<b>25</b>
<b>CODE 19 SIP STATUS CODE CONSTANTS.....</b>	<b>27</b>
<b>CODE 20 NON-STANDARD PARAMETER DECLARATION.....</b>	<b>28</b>
<b>CODE 21 RECEIVE DATA BUFFER DECLARATION.....</b>	<b>33</b>
<b>CODE 22 TRANSMIT DATA BUFFER DECLARATION.....</b>	<b>34</b>
<b>CODE 23 SAMPLE: STATELESS RESPONSE.....</b>	<b>39</b>
<b>CODE 24 SAMPLE: STATELESS RESPONSE.....</b>	<b>39</b>
<b>CODE 25 STATELESS REDIRECTION.....</b>	<b>40</b>
<b>CODE 26 SENDING STATELESS REQUEST.....</b>	<b>41</b>
<b>CODE 27 STATELESS FORWARDING.....</b>	<b>42</b>
<b>CODE 28 SENDING STATEFULL RESPONSE.....</b>	<b>47</b>
<b>CODE 29 SENDING STATEFULL RESPONSE.....</b>	<b>47</b>
<b>CODE 30 SENDING REQUEST STATEFULLY.....</b>	<b>48</b>
<b>CODE 31 SENDING REQUEST STATEFULLY.....</b>	<b>48</b>
<b>CODE 32 STATEFULL FORWARDING.....</b>	<b>49</b>
<b>CODE 33 CLIENT AUTHORIZATION EXAMPLE.....</b>	<b>52</b>
<b>CODE 34 CREATING DIALOG FOR INCOMING INVITE.....</b>	<b>61</b>
<b>CODE 35 ANSWERING DIALOG.....</b>	<b>62</b>
<b>CODE 36 PROCESSING CANCEL REQUEST.....</b>	<b>63</b>
<b>CODE 37 PROCESSING ACK REQUEST.....</b>	<b>63</b>
<b>CODE 38 CREATING OUTGOING DIALOG.....</b>	<b>64</b>
<b>CODE 39 RECEIVING RESPONSE IN DIALOG.....</b>	<b>64</b>
<b>CODE 40 SENDING ACK REQUEST.....</b>	<b>65</b>

## Chapter 1: General Design

## 1.1 Architecture

### 1.1.1 Communication Diagram

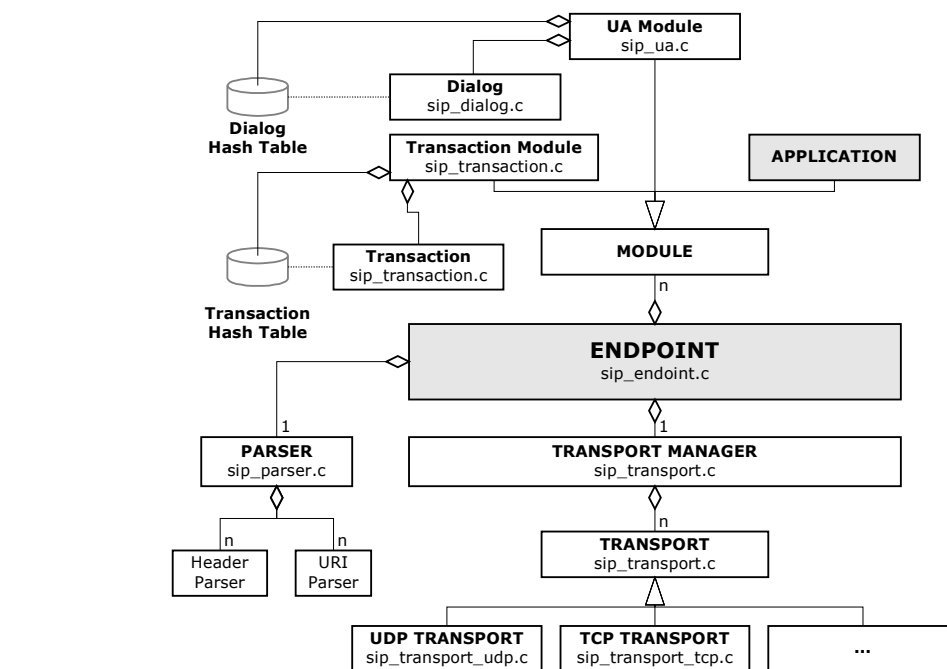
The following diagram shows how (SIP) messages are passed back and forth among PJSIP components.



### Figure 1 Collaboration Diagram

### 1.1.2 Class Diagram

The following diagram shows the “class diagram”.



### Figure 2 Class Diagram



## 1.2 Module

Module can be added to PJSIP to extend its functionality. Module is mainly used to register callback to receive SIP messages from endpoint.

### 1.2.1 Module Declaration

A module is declared as follows.

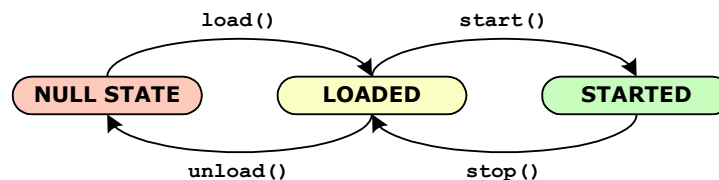
```
struct pjsip_module
{
    pj_str_t      name;           // Module name.
    int           id;             // Module ID, set by endpt
    int           priority;       // Priority
    void          *data;          // User data.

    int           method_cnt;     // Nb.of supported methods
    const pjsip_method *methods[8]; // Array supported methods

    pj_status_t (*load)          (pjsip_endpoint *endpt); // Called to load the mod.
    pj_status_t (*start)         (void);                  // Called to start.
    pj_status_t (*stop)          (void);                  // Called top stop.
    pj_status_t (*unload)        (void);                  // Called before unload
    pj_bool_t  (*on_rx_request)  (pjsip_rx_data *rdata); // Called on rx request
    pj_bool_t  (*on_rx_response) (pjsip_rx_data *rdata); // Called on rx response
    void        (*on_tsx_state)  (pjsip_transaction *tsx, // Called on transaction
                                pjsip_event *event);      // state changed
};
```

**Code 1 Module Declaration**

The four function pointers `load`, `start`, `stop`, and `unload` are called by endpoint to control the module state. The following diagram shows the module's state lifetime.



**Figure 3 Module State Diagram**

The `on_rx_request()` and `on_rx_response()` function pointers are the primary means for the module to receive SIP messages from endpoint (`pjsip_endpt`) or from other modules. The return value of these callbacks is important. If a callback has returned non-zero (i.e. true condition), it semantically means that the module has taken care the message; in this case, the endpoint will stop distributing the message to other modules.

The `on_tsx_state()` function pointer is used to receive notification every time a transaction state has changed. It is different from `on_rx_request()` and `on_rx_response()` callback because it's only called when transaction state has actually changed, which means it's not called for example when transaction receives 180/Ringing response after 100/Trying response. It also means that this callback may be called for other non-message-arrival related events (e.g. message transmissions, timer timeout event, or transport error event). More information about this callback will be described in next section 1.2.4 "Transaction User and State Callback".

## 1.2.2 Module Priorities

Module priority specifies the order of which modules are called first to process the callback. Module with higher priority (i.e. lower priority *number*) gets called first.

The values below are guidelines to set module priority.

```
enum pjsip_module_priority
{
    PJSIP_MOD_PRIORITY_TSX_LAYER      = 4,    // Transaction layer.
    PJSIP_MOD_PRIORITY_UA_PROXY_LAYER = 16,    // UA or proxy layer
    PJSIP_MOD_PRIORITY_APPLICATION    = 32,    // Application has lowest priority.
};
```

**Code 2 Module Priorities**



**Note:** remember that lower priority *number* means higher priority!

Applications MUST set its priority *number* to PJSIP\_MOD\_PRIORITY\_APPLICATION or higher value, to make sure that it receives the message last, after the message has been processed by the relevant modules. A module that wishes to receive ALL messages (e.g. for decryption purpose) SHOULD set its priority *number* lower than PJSIP\_MOD\_PRIORITY\_TSX\_LAYER.

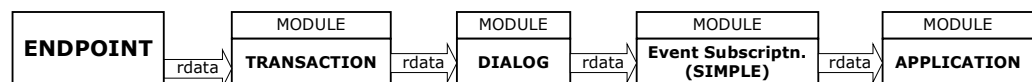
## 1.2.3 Message Processing by Modules

When incoming message arrives, it is represented as receive message buffer (`struct pjsip_rx_data`, see section 4.1 "Receive Data Buffer"). Transport manager parses the message, put the parsed data structures in the receive message buffer, and passes the message to the endpoint.

The endpoint distributes the receive message buffer to each registered module by calling `on_rx_request()` or `on_rx_response()` callback, starting from module with highest priority (i.e. lowest priority *number*) until one of them returns non-zero. When one of the module has returned non-zero, endpoint stops distributing the message to the remaining of the modules, because it assumes that the module has taken care about the processing of the message.

The module which returns non-zero on the callback itself may further distribute the message to other modules. For example, the transaction module, upon receiving matching message, will process the message then distributes the message to its transaction user, which in itself must be a module too. The transaction passes the message to the transaction user (i.e. a module) by calling `on_rx_request()` or `on_rx_response()` callback of that module, after setting the transaction field in the receive message buffer so that the transaction user module can distinguish between messages that are outside transactions and messages that are inside a transaction.

The following diagram shows an example of how modules may cascadelly call other modules.



**Figure 4 Cascade Module Callback**

### 1.2.4 Transaction User and State Callback

A special callback in the module definition (`on_tsx_state`) is used to receive notification from a particular transaction when transaction state has changed. This callback is unique because transaction state may change because of non-message related events (e.g. timer timeout and transport error), and this callback is called for any changes in the transaction's state (e.g. because of transmission or receipt of messages, timeout, transport error, etc).

This callback will only be called after the module has been registered as transaction user for a particular transaction. Only one transaction user is allowed per transaction. Transaction user can be set to transaction on per transaction basis.

Normally when a transaction is created within a dialog, then the transaction user will be the UA layer on behalf of a particular dialog. But when applications work on top of the transaction layer directly, they may set themselves as the transaction user.

### 1.2.5 Module Specific Data

Some PJSIP components have a container where modules can put module specific data in that component. This container is named as `mod_data` by convention, and is an array of pointer to void, which is indexed by the module ID.

For example, an incoming packet buffer (`pjsip_rx_data`) has the following declaration for module specific data container:

```
struct pjsip_rx_data
{
    ...
    struct {
        void *mod_data[PJSIP_MAX_MODULE];
    } endpt_info;
};
```

**Code 3 Module Specific Data**

When an incoming packet buffer (`pjsip_rx_data`) is passed around to modules, a module can put module specific data in the appropriate index in `mod_data`, so that the value can be picked up later by the module or by application. For example, the transaction layer will put the matching transaction instance in the `mod_data`, and user agent layer will put the matching dialog instance in the `mod_data` too. Application can retrieve the value calling `pjsip_rdata_get_tsx()` or `pjsip_rdata_get_dlg()`, which is a simple array lookup function as follows:

```
// This code can be found in sip_transaction.c

static pjsip_module mod_tsx_layer;

pjsip_transaction *pjsip_rdata_get_tsx(pjsip_rx_data *rdata)
{
    return rdata->endpt_info.mod_data[mod_tsx_layer.id];
}
```

**Code 4 Accessing Module Specific Data**

### 1.2.6 Callback Summary

The following table summarizes the occurrence of an event and the triggering of particular callbacks. The `on_tsx_state()` callback will of course only be called when application has chosen to process a request statefully.

Event	<code>on_rx_request()</code> <b>or</b> <code>on_rx_response()</code>	<code>on_tsx_state()</code>
Receipt of new requests or responses	Called	Called
Receipt retransmissions of requests or responses.	Called ONLY when priority number is lower than transaction layer <sup>1</sup>	Not Called
Transmission of new requests or responses.	Not Called	Called
Retransmissions of requests or responses.	Not Called	Not Called
Transaction timeout	Not Called	Called
Other transaction failure events (e.g. DNS query failure, transport failure)	Not Called	Called

**Figure 5 Callback Summary**

### 1.2.7 Sample Callback Requirements for Applications

The following table summarizes the requirements for the callbacks for each logical type of applications. Note that any of these logical applications may co-exist in a single physical/executable program, and practically it's application's decision to invoke the appropriate logical functionalities or whether to work statefull or statelessly. This decision is made on per request basis.

Application	Requirements
<b>Stateless Proxies</b>	Stateless proxies need to receive: 1) all incoming <b>requests</b> . 2) al incoming <b>responses</b> .
<b>Statefull Proxies</b>	Statefull proxies need to receive: 1) new incoming requests (i.e. that are not attached to any transactions). 2) all responses received by a transaction, and 3) any other transaction events (e.g. DNS failure, timeout, transport error).
<b>Statefull/less Registrar Server</b>	Registrar servers need to receive: 1) all incoming REGISTER requests.
<b>UA</b>	Typical UA applications need to receive: 1) incoming requests that are not attached to any transactions. 2) all requests that belong to the dialog. 3) all incoming responses associated with the dialog and any other transaction events (e.g. timeout, transport error).

**Figure 6 Sample Callback Requirements**

<sup>1</sup> This is because the matching transaction prevents the message from being distributed further (by returning `PJ_TRUE`) and it also does NOT call TU callback upon receiving retransmissions.

### 1.2.8 Sample Callback Diagrams

#### Incoming Message Outside Transaction and Outside Dialog

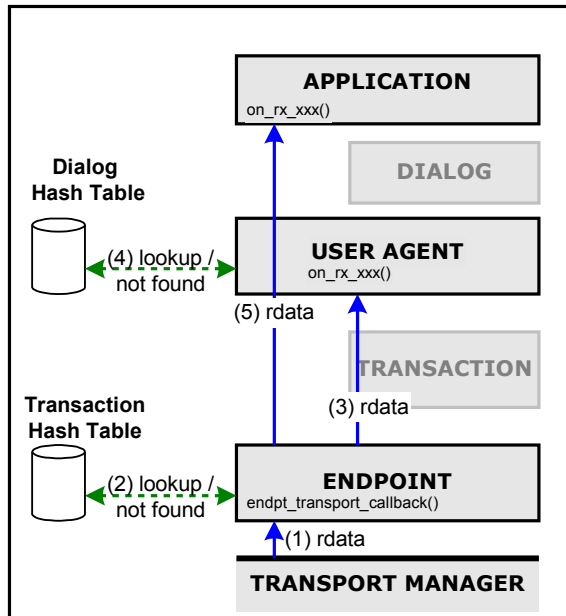


Figure 7 Processing of Incoming Message Outside Transaction/Dialog

The processing is as follows:

- 1) Transport manager (pjsip\_tpmgr) passes all received messages to endpoint (after parsing the message).
- 2) Endpoint (pjsip\_endpt) distributes the message to all registered callbacks. First in the callback list is transaction layer. Endpoint lookup the message in transaction table, and couldn't find a matching transaction.
- 3) Endpoint distributes the message to next callback in the list, which is user agent.
- 4) User agent lookup the message in dialog's hash table and couldn't find matching dialog.
- 5) Endpoint continues distributing the message to next registered callbacks until it reaches application. Application processes the message (e.g. create UAS transaction, or proxy the request, or create dialog, etc.)

#### Incoming Message Inside Transaction

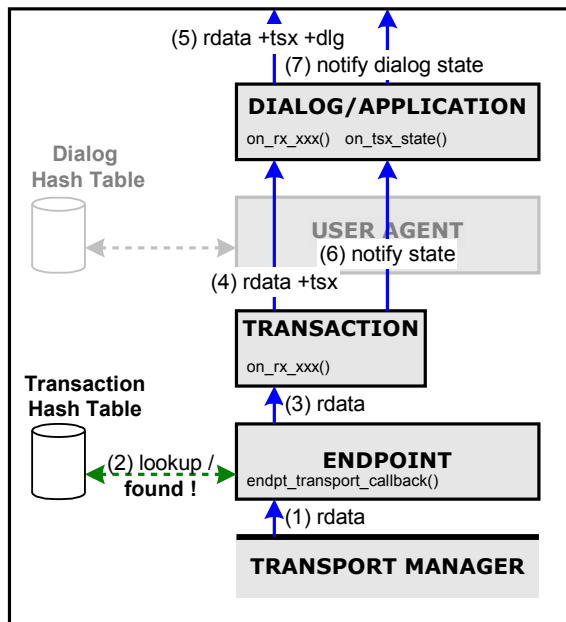
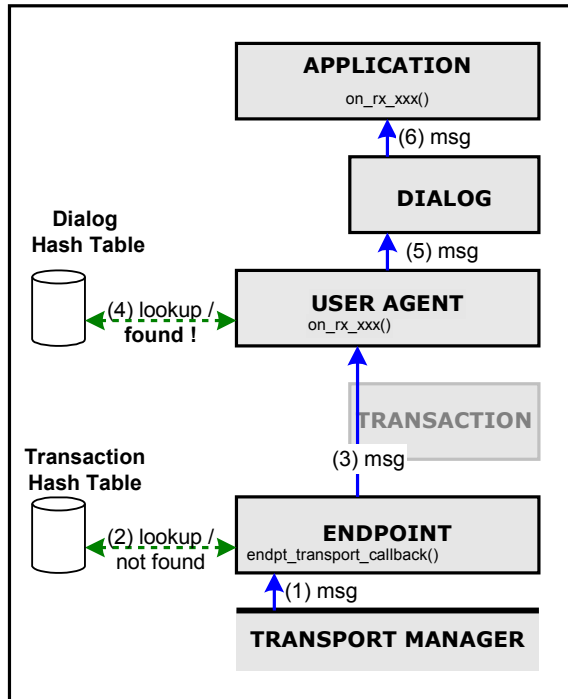


Figure 8 Processing of Incoming Message Inside Transaction

The processing is as follows:

- 1) Transport manager (pjsip\_tpmgr) passes all received messages to endpoint (after parsing the message).
- 2) Endpoint (pjsip\_endpt) distributes the message to all registered callbacks. First in the callback list is transaction layer. Endpoint lookup the message in transaction table, and **found** a matching transaction.
- 3) Endpoint distributes the message to the transaction. Because transaction's callback returns non-zero, endpoint does not distribute the message to the rest of the registered callbacks.
- 4) The transaction processes the response (e.g. updates the FSM). If the message is a retransmission, the processing stops here. Otherwise transaction then passes the message to it's transaction user (TU), which can be a dialog or application.
- 5) If the TU is a dialog, the dialog processes the response then pass the response to it's dialog user (DU, e.g. application).
- 6) If the arrival of the message has changed transaction's state, transaction will notify it's TU about the new state.
- 7) If TU is a dialog, it may further notify application about dialog's state changed.

### Incoming Message Inside Dialog but Outside Transaction



The processing is as follows:

- 1) Transport manager (pjsip\_tpmgr) passes all received messages to endpoint (after parsing the message).
- 2) Endpoint (pjsip\_endpt) distributes the message to all registered callbacks. First in the callback list is transaction layer. Endpoint lookup the message in transaction table, and couldn't find a matching transaction.
- 3) Endpoint distributes the message to next modules in the list, until it reaches user agent module.
- 4) The user agent module looks-up the owning of the message in dialog's hash table and found a matching dialog.
- 5) The user agent module passes the message to the appropriate dialog.
- 6) The dialog processes the message, updates its state etc, and notify the application.

**Figure 9 Processing of Incoming Message Inside Dialog but Outside Transaction**

## 1.3 Module Management

Modules are managed by PJSIP's endpoint (`pjsip_endpoint`). Application MUST register each module manually to endpoint so that it can be recognized by the stack. Application can register or unregister module at any time during application's life-time, although it is recommended to register module only during startup and to unregister module only during application exit.



All PJSIP modules can be registered/unregistered dynamically at anytime during application's lifetime. However by doing so, it may severely change the handling of messages. For example, when transaction module is unregistered, application may receive strayed responses that are no longer associated with any transactions.

### 1.3.1 Module Management API



```

pj_status_t pjsip_endpt_register_module(    pjsip_endpoint *endpt,
                                             pjsip_module *module );
  
```

Register a module to the endpoint. The endpoint will then call the load and start function in the module to properly initialize the module, and assign a unique module ID for the module.



```

pj_status_t pjsip_endpt_unregister_module( pjsip_endpoint *endpt,
                                             pjsip_module *module );
  
```

Unregister a module from the endpoint. The endpoint will then call the stop and unload function in the module to properly shutdown the module.

## Chapter 2: Message Elements

### 2.1 Uniform Resource Indicator (URI)

The Uniform Resource Indicator (URI) in PJSIP is modeled pretty much in object oriented manner (or some may argue it's object based, not object oriented). Because of this, URI can be treated uniformly by the stack, and new types URI can be introduced quite easily.

#### 2.1.1 URI "Class Diagram"

The following diagram shows show the URI objects are designed.

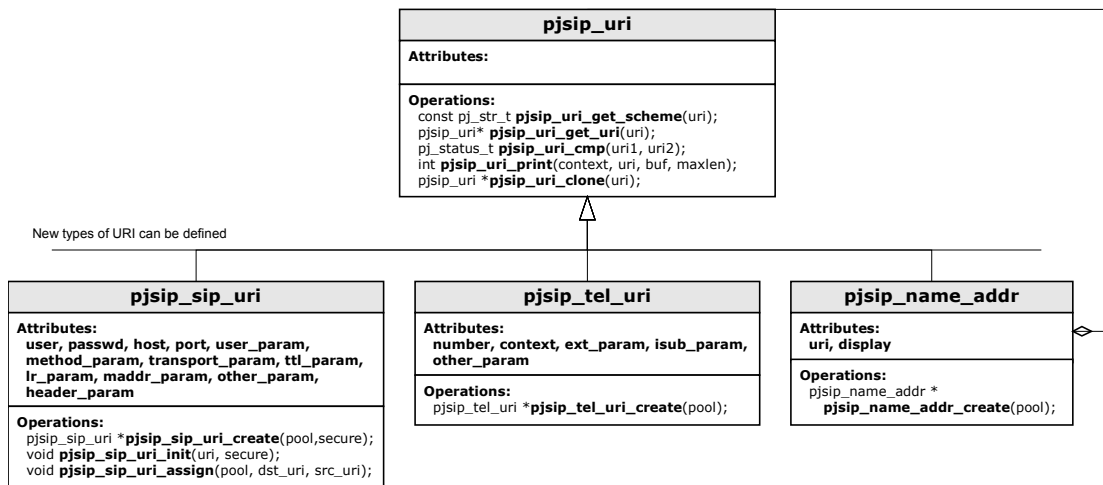


Figure 10 URI "Class Diagram"

More information on each objects will be described in next sections.

#### 2.1.2 URI Context

URI context specifies where the URI is being used (e.g. in request line, in From/To header, etc.). The context specifies what URI elements are allowed to appear in that context. For example, transport parameter is not allowed to appear in From/To header, etc.

In PJSIP, the context must be specified when printing the URI to a buffer and when comparing two URIs. In this case, the parts of URI that is not allowed to appear in the specified context will be ignored during printing and comparison process.

```

enum pjsip_uri_context_e
{
    PJSIP_URI_IN_REQ_URI,           // The URI is in Request URI.
    PJSIP_URI_IN_FROMTO_HDR,       // The URI is in From/To header.
    PJSIP_URI_IN_CONTACT_HDR,      // The URI is in Contact header.
    PJSIP_URI_IN_ROUTING_HDR,      // The URI is in Route/Record-Route header.
    PJSIP_URI_IN_OTHER,            // Other context (web page, business card, etc.)
};
    
```

Code 5 URI Context

### 2.1.3 Base URI

The `pjsip_uri` structure contains property that is shared by all types of URI. Because of this, all types of URI can be type-casted to `pjsip_uri` and manipulated uniformly.

```
struct pjsip_uri
{
    pjsip_uri_vptr *vptr;
};
```

**Code 6 Generic URI Declaration**

The `pjsip_uri_vptr` specifies "virtual" function table, which members will be defined by each type of URI. Application is discouraged from calling these function pointers directly; instead it is recommended to use the URI API because they are more readable (and it saves some typings too).

```
struct pjsip_uri_vptr
{
    const pj_str_t* (*p_get_scheme) ( const pjsip_uri *uri);
    pjsip_uri*      (*p_get_uri)    ( pjsip_uri *uri);
    int             (*p_print)      ( pjsip_uri_context_e context,
                                     const pjsip_uri *uri,
                                     char *buf, pj_size_t size);

    pj_status_t     (*p_compare)    ( pjsip_uri_context_e context,
                                     const pjsip_uri *uri1, const pjsip_uri *uri2);
    pjsip_uri *     (*p_clone)     ( pj_pool_t *pool, const pjsip_uri *uri);
};
```

**Code 7 URI Virtual Function Table**

The URI functions below can be applied for all types of URI objects. These functions normally are implemented as inline functions which call the corresponding function pointer in virtual function table of the URI.

```
const pj_str_t* pjsip_uri_get_scheme( const pjsip_uri *uri );
    Get the URI scheme string (e.g. "sip", "sips", "tel", etc.).
```

```
pjsip_uri* pjsip_uri_get_uri( pjsip_uri *uri );
    Get the URI object. Normally all URI objects will return itself except name
    address which will return the URI inside the name address object.
```

```
pj_status_t pjsip_uri_cmp( pjsip_uri_context_e context,
                           const pjsip_uri *uri1,
                           const pjsip_uri *uri2);
    Compare uri1 and uri2 according to the specified context. Parameters
    which are not allowed to appear in the specified context will be ignored in
    the comparison. It will return PJ_SUCCESS is both URIs are equal.
```

```
int pjsip_uri_print(    pjsip_uri_context_e context,
                        const pjsip_uri *uri,
                        char *buffer,
                        pj_size_t max_size);
    Print uri to the specified buffer according to the specified context.
    Parameters which are not allowed to appear in the specified context will
    not be included in the printing.
```



```
pjsip_uri* pjsip_uri_clone( pj_pool_t *pool, const pjsip_uri *uri );
```

Create a deep clone of *uri* using the specified pool.

### 2.1.4 SIP and SIPS URI

The structure `pjsip_sip_uri` represents SIP and SIPS URI scheme. It is declared in `<pjsip/sip_uri.h>`.

```
struct pjsip_sip_uri
{
    pjsip_uri_vptr *vp_ptr;           // Pointer to virtual function table.
    pj_str_t        user;             // Optional user part.
    pj_str_t        passwd;          // Optional password part.
    pj_str_t        host;             // Host part, always exists.
    int             port;             // Optional port number, or zero.
    pj_str_t        user_param;       // Optional user parameter
    pj_str_t        method_param;     // Optional method parameter.
    pj_str_t        transport_param;  // Optional transport parameter.
    int             ttl_param;        // Optional TTL param, or -1.
    int             lr_param;         // Optional loose routing param, or 0
    pj_str_t        maddr_param;      // Optional maddr param
    pjsip_param     other_param;      // Other parameters as list.
    pjsip_param     header_param;     // Optional header parameters as list.
};
```

**Code 8 SIP URI Declaration**

The following functions are specific to SIP/SIPS URI objects. In addition to these functions, application can also use the functions described in previous section "" to manipulate SIP and SIPS URI too.

```
pjsip_sip_uri* pjsip_sip_uri_create( pj_pool_t *pool, pj_bool_t secure );
```

Create a new SIP URL using the specified *pool*. If the *secure* flag is set to non-zero, then SIPS URL will be created. This function will set *vp\_ptr* member of the URL to SIP or SIPS *vp\_ptr* and set all other members to blank value.

```
void pjsip_sip_uri_init( pjsip_sip_uri *url, pj_bool_t secure );
```

Initialize a SIP URL structure.

```
void pjsip_sip_uri_assign( pj_pool_t *pool,
                          pjsip_sip_uri *url,
                          const pjsip_sip_uri *rhs );
```

Perform deep copy of *rhs* to *url*.

### 2.1.5 Tel URI

The structure `pjsip_tel_uri` represents **tel:** URL. It is declared in `<pjsip/sip_tel_uri.h>`.

```
struct pjsip_tel_uri
{
```

```

pjsip_uri_vptr *vptr;           // Pointer to virtual function table.
pj_str_t      number;           // Global or local phone number
pj_str_t      context;          // Phone context (for local number).
pj_str_t      ext_param;        // Extension param.
pj_str_t      isub_param;       // ISDN sub-address param.
pjsip_param    other_param;     // Other parameters.
};

```

**Code 9 TEL URI Declaration**

The functions below are specific to TEL URI. In addition to these functions, application can also use the functions described in previous section "" for TEL URI too.

```

pjsip_tel_uri* pjsip_tel_uri_create( pj_pool_t *pool );
    Create a new tel: URI.

```

```

int pjsip_tel_nb_cmp( const pj_str_t *nb1, const pj_str_t *nb2 );
    This utility function compares two telephone numbers for equality,
    according to rules specified in RFC 3966 (about tel: URI). It recognizes
    global and local numbers, and it ignores visual separators during the
    comparison.

```

### 2.1.6 Name Address

A name address (`pjsip_name_addr`) does not really define a new type of URI, but rather encapsulates existing URI (e.g. SIP URI) and adds display name.

```

struct pjsip_name_addr
{
    pjsip_uri_vptr *vptr;           // Pointer to virtual function table.
    pj_str_t      display;          // Display name.
    pjsip_uri      *uri;            // The URI.
};

```

**Code 10 Name Address Declaration**

The following functions are specific to name address URI object. In addition to these functions, application can also use the functions described in Section "" for name address object too.

```

pjsip_name_addr* pjsip_name_addr_create( pj_pool_t *pool );
    Create a new name address. This will set initialize the virtual function table
    pointer, set blank display name and set the uri member to NULL.

void pjsip_name_addr_assign( pj_pool_t *pool,
                             pjsip_name_addr *name_addr,
                             const pjsip_name_addr *rhs );
    Copy rhs to name_addr.

```

### 2.1.7 Sample URI Manipulation Program

```

#include <pjlib.h>
#include <pjsip_core.h>
#include <stdlib.h>           // exit()

```

```

static pj_caching_pool cp;

static void my_error_exit(const char *title, pj_status_t errcode)
{
    char errbuf[80];
    pjsip_strerror(errcode, errbuf, sizeof(errbuf));
    PJ_LOG(3, ("main", "%s: %s", title, errbuf));
    exit(1);
}

static void my_init_pjlib(void)
{
    pj_status_t status;
    // Init PJLIB
    status = pj_init();
    if (status != PJ_SUCCESS) my_error_exit("pj_init() error", status);
    // Init caching pool factory.
    pj_caching_pool_init( &cp, &pj_pool_factory_default_policy, 0);
}

static void my_print_uri( const char *title, pjsip_uri *uri )
{
    char buf[80];
    int len;

    len = pjsip_uri_print( PJSIP_URI_IN_OTHER, uri, buf, sizeof(buf)-1);
    if (len < 0)
        my_error_exit("Not enough buffer to print URI", -1);

    buf[len] = '\0';
    PJ_LOG(3, ("main", "%s: %s", title, buf));
}

int main()
{
    pj_pool_t *pool;
    pjsip_name_addr *name_addr;
    pjsip_sip_uri *sip_uri;

    // Init PJLIB
    my_init_pjlib();

    // Create pool to allocate memory
    pool = pj_pool_create(&cp.factory, "mypool", 4000, 4000, NULL);
    if (!pool) my_error_exit("Unable to create pool", PJ_ENOMEM);

    // Create and initialize a SIP URI instance
    sip_uri = pjsip_sip_uri_create(pool, PJ_FALSE);
    sip_uri->user = pj_str("alice");
    sip_uri->host = pj_str("sip.example.com");

    my_print_uri("The SIP URI is", (pjsip_uri*)sip_uri);

    // Create a name address to put the SIP URI
    name_addr = pjsip_name_addr_create(pool);
    name_addr->uri = (pjsip_uri*) sip_uri;
    name_addr->display = "Alice Cooper";

    my_print_uri("The name address is", (pjsip_uri*)name_addr);
    // Done
}

```

**Code 11 Sample URI Manipulation Program**

## 2.2 SIP Methods

### 2.2.1 SIP Method Representation (`pjsip_method`)

The SIP method representation in PJSIP is also extensible; it can support new methods without needing to recompile the library.

```
struct pjsip_method
{
    pjsip_method_e id;        // Method ID, from pjsip_method_e.
    pj_str_t        name;     // Method name, which will always contain the method string.
};
```

**Code 12 SIP Method Declaration**

PJSIP core library declares only methods that are specified in core SIP standard (RFC 3261). For these core methods, the `id` field of `pjsip_method` will contain the appropriate value from the following enumeration:

```
enum pjsip_method_e
{
    PJSIP_INVITE_METHOD,
    PJSIP_CANCEL_METHOD,
    PJSIP_ACK_METHOD,
    PJSIP_BYE_METHOD,
    PJSIP_REGISTER_METHOD,
    PJSIP_OPTIONS_METHOD,

    PJSIP_OTHER_METHOD,
};
```

**Code 13 SIP Method ID**

For methods not specified in the enumeration, the `id` field of `pjsip_method` will contain `PJSIP_OTHER_METHOD` value. In this case, application must inspect the `name` field of `pjsip_method` to know the actual method.

### 2.2.2 SIP Method API

The following functions can be used to manipulate PJSIP's SIP method objects.

```
void pjsip_method_init(    pjsip_method *method, pj_pool_t *pool,
                          const pj_str_t *method_name );
    Initialize method from string. This will initialize the id of the method field
    to the correct value.

void pjsip_method_init_np( pjsip_method *method,
                          pj_str_t *method_name );
    Initialize method from method_name string without duplicating the string
    (np stands for no pool). The id field will be initialize accordingly.

void pjsip_method_set(    pjsip_method *method,
                          pjsip_method_id_e method_id );
    Initialize method from the method ID enumeration. The name field will be
    initialized accordingly.

void pjsip_method_copy(    pj_pool_t *pool,
                          pjsip_method *method,
                          const pjsip_method *rhs );
```

Copy *rhs* to *method*.

```
int pjsip_method_cmp( const pjsip_method *method1,
                     const pjsip_method *method2 );
```

Compare *method1* to *method2* for equality. This function returns zero if both methods are equal, and (-1) or (+1) if *method1* is less or greater than *method2* respectively.

## 2.3 Header Fields

All header fields in PJSIP share common header properties such as header type, name, short name, and virtual function table. Because of this, all header fields can be treated uniformly by the stack.

### 2.3.1 Header "Class Diagram"

The following diagram shows the snippet of PJSIP header "class diagram". There are more headers than the ones shown in the diagram; PJSIP library implements ALL headers that are specified in the core SIP specification (RFC 3261). Other headers will be implemented in the corresponding PJSIP extension module.

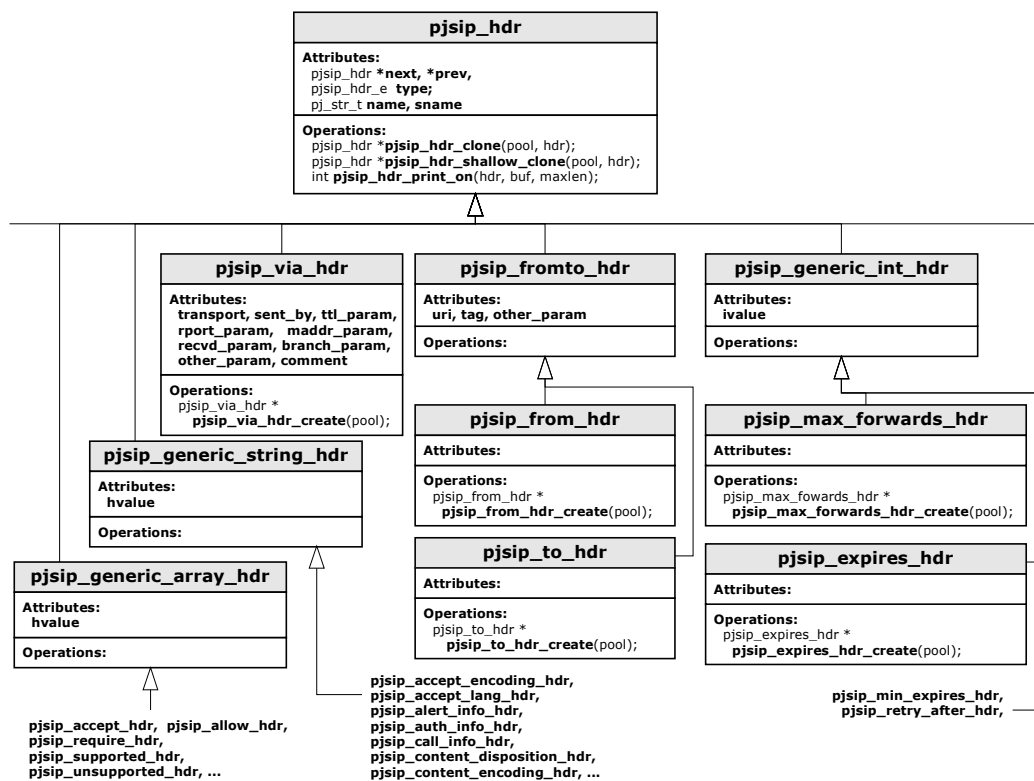


Figure 11 Header "Class Diagram"

As seen in the "class diagram", each of the specific header normally only provide one function that is specific for that particular header, i.e. function to create the instance of the header.

### 2.3.2 Header Structure

To make sure that header fields contain common header properties and those properties are in the correct and same memory layout, the header declaration must call `PJSIP_DECL_HDR_MEMBER` macro as the first member field of the header, specifying the header name as argument to the macro.

```
#define PJSIP_DECL_HDR_MEMBER(hdr)      \
    /** List members. */                 \
    PJ_DECL_LIST_MEMBER(hdr);           \
    /** Header type */                   \
    pjsip_hdr_e type;                    \
    /** Header name. */                  \
    pj_str_t name;                       \
    /** Header short name version. */    \
    pj_str_t sname;                      \
    /** Virtual function table. */       \
    pjsip_hdr_vptr *vptr
```

**Code 14 Generic Header Declaration**

PJSIP defines `pjsip_hdr` structure, which contains common properties shared by all header fields. Because of this, all header fields can be typecasted to `pjsip_hdr` so that they can be manipulated uniformly.

```
struct pjsip_hdr
{
    PJSIP_DECL_HDR_MEMBER(struct pjsip_hdr);
};
```

**Code 15 Generic Header Declaration**

### 2.3.3 Common Header Functions

The `pjsip_hdr_vptr` specifies “virtual” function table, which implementation is provided by each header types. The table contains pointer to functions as follows:

```
struct pjsip_hdr_vptr
{
    pjsip_hdr *(*clone)      ( pj_pool_t *pool, const pjsip_hdr *hdr );
    pjsip_hdr *(*shallow_clone)( pj_pool_t *pool, const pjsip_hdr *hdr );
    int      (*print_on)    ( pjsip_hdr *hdr, char *buf, pj_size_t len );
};
```

**Code 16 Header Virtual Function Table**

Although application can freely call the function pointers in the `pjsip_hdr_vptr` directly, it is recommended that it uses the following header APIs instead, because they will make the program more readable.

```
pjsip_hdr *pjsip_hdr_clone(    pj_pool_t *pool,
                               const pjsip_hdr *hdr );
```

Perform deep clone of *hdr* header.

```
pjsip_hdr *pjsip_hdr_shallow_clone( pj_pool_t *pool,
                                     const pjsip_hdr *hdr );
```

Perform shallow clone of *hdr* header. A shallow cloning creates a new exact copy of the specified header field, however most of its value will still point to the values in the original header. Normally shallow clone is just a simple `memcpy()` from the original header to a new header, therefore it's expected that this operation is faster than deep cloning.

However, care must be taken when shallow cloning headers. It must be understood that the new header still shares common pointers to the values in the old header. Therefore, when the pool containing the original header is destroyed, the new header will be rendered invalid too although the new header was shallow-cloned using different memory pool. Or if some values in the original header was modified, then the corresponding values in the shallow-cloned header will be modified too.

Despite of this, shallow cloning is used widely in the library. For example, a dialog has some headers which values are more or less persistent during the session (e.g. From, To, Call-Id, Route, and Contact). When creating a request, the dialog can just shallow-clone these headers (instead of performing full cloning) and put them in the request message.

```
int pjsip_hdr_print_on(pjsip_hdr *hdr,
                      char *buf,
                      pj_size_t max_size);
```

Print the specified header to a buffer (e.g. before transmission). This function returns the number of bytes printed to the buffer, or -1 when the buffer is overflow.

### 2.3.4 Supported Header Fields

The "standard" PJSIP header fields are declared in <pjsip/sip\_msg.h>. Other header fields may be declared in header files that implement specific functionalities or SIP extensions (e.g. headers used by SIMPLE extension, etc.).

Each header field normally only defines one specific API for manipulating them, i.e. the function to create that specific header field. Other APIs are exported through the virtual function table.

The APIs to create individual header fields are by convention named after the header field name and followed by `_create()` suffix. For example, call function `pjsip_via_hdr_create()` to create an instance of `pjsip_via_hdr` header.

Please refer to <pjsip/sip\_msg.h> for complete list of header fields defined by PJSIP core.

### 2.3.5 Header Array Elements

A lot of SIP headers (e.g. Require, Contact, Via, etc.) can be grouped together as a single header field and separated by comma. Example:

```
Contact: <sip:alice@sip.example.com>;q=1.0, <tel:+442081234567>;q=0.5
Via: SIP/2.0/UDP proxy1.example.com;branch=z9hG4bK87asdk7, SIP/2.0/UDP
    proxy2.example.com;branch=z9hG4bK77asjd
```



**NOTE:** PJSIP does not support representing array elements in a header for complex header types (e.g. Contact, Via, Route, Record-Route). Simple string array however is supported (e.g. Require, Supported, etc.).

When the parser encounters such arrays in headers, it will split the array into individual headers while maintaining their order of appearance. So for the example above, the parser will modify the message to:

```
Contact: <sip:alice@sip.example.com>;q=1.0
Contact: <tel:+442081234567>;q=0.5
Via: SIP/2.0/UDP proxy1.example.com;branch=z9hG4bK87asdk7
Via: SIP/2.0/UDP proxy2.example.com;branch=z9hG4bK77asjd
```

The SIP standard specifies that there should NOT be any difference in the processing of message containing either kind of header representations. So we believe that the removal of header array support will not limit the functionality of PJSIP at all.

The reason why we impose this limitation is because based on our experience, the removal of header array support greatly simplifies processing of headers. If header array were supported, then application not only must inspect all headers, it also has to inspect some headers to see if they contain arrays. With the removal of array support, application only has to inspect the main header list in the message.

## 2.4 Message Body (pjsip\_msg\_body)

SIP message body is represented with `pjsip_msg_body` structure in PJSIP. This structure is declared in `<pjsip/sip_msg.h>`.

```
struct pjsip_msg_body
{
    /** MIME content type.
     * For incoming messages, the parser will fill in this member with the
     * content type found in Content-Type header.
     *
     * For outgoing messages, application must fill in this member with
     * appropriate value, because the stack will generate Content-Type header
     * based on the value specified here.
     */
    pjsip_media_type content_type;

    /** Pointer to buffer which holds the message body data.
     * For incoming messages, the parser will fill in this member with the
     * pointer to the body string.
     *
     * When sending outgoing message, this member doesn't need to point to the
     * actual message body string. It can be assigned with arbitrary pointer,
     * because the value will only need to be understood by the print_body()
     * function. The stack itself will not try to interpret this value, but
     * instead will always call the print_body() whenever it needs to get the
     * actual body string.
     */
    void *data;

    /** The length of the data.
     * For incoming messages, the parser will fill in this member with the
     * actual length of message body.
     *
     * When sending outgoing message, again just like the "data" member, the
     * "len" member doesn't need to point to the actual length of the body
     * string.
     */
    unsigned len;

    /** Pointer to function to print this message body.
     * Application must set a proper function here when sending outgoing
     * message.
     *
     * @param msg_body      This structure itself.
     * @param buf           The buffer.
     * @param size          The buffer size.
     *
     * @return              The length of the string printed, or -1 if there is
     *                      not enough space in the buffer to print the whole
     *                      message body.
     */
    int (*print_body)    ( struct pjsip_msg_body *msg_body,
```





```

char *buf, pj_size_t size );

/** Pointer to function to clone the data in this message body.
 */
void* (*clone)          ( struct pjsip_msg_body *msg_body,
                          void *data, unsigned len );
};

```

**Code 17 Message Body Declaration**

## 2.5 Message (pjsip\_msg)

Both request and response message in PJSIP are represented with `pjsip_msg` structure in `<pjsip/sip_msg.h>`. The following code snippet shows the declaration of `pjsip_msg` along with other supporting structures.

```

enum pjsip_msg_type_e
{
    PJSIP_REQUEST_MSG,          // Indicates request message.
    PJSIP_RESPONSE_MSG,        // Indicates response message.
};

struct pjsip_request_line
{
    pjsip_method  method;       // Method for this request line.
    pjsip_uri     *uri;         // URI for this request line.
};

struct pjsip_status_line
{
    int           code;         // Status code.
    pj_str_t     reason;       // Reason string.
};

struct pjsip_msg
{
    /** Message type (ie request or response). */
    pjsip_msg_type_e  type;

    /** The first line of the message can be either request line for request
     * messages, or status line for response messages. It is represented here
     * as a union.
     */
    union
    {
        /** Request Line. */
        struct pjsip_request_line  req;

        /** Status Line. */
        struct pjsip_status_line   status;
    } line;

    /** List of message headers. */
    pjsip_hdr hdr;

    /** Pointer to message body, or NULL if no message body is attached to
     * this message.
     */
    pjsip_msg_body *body;
};

```

**Code 18 SIP Message Declaration**

The following are APIs that are provided for manipulating SIP message objects.

```
pjsip_msg* pjsip_msg_create( pj_pool_t *pool,
```

```
pjsip_msg_type_e type);
```

Create a request or response message according to the *type*.

```
pjsip_hdr* pjsip_msg_find_hdr(    pjsip_msg *msg,
                                   pjsip_hdr_e hdr_type,
                                   pjsip_hdr *start);
```

Find header in the *msg* which has the specified *type*, searching from (and including) the specified *start* position in the header list. If *start* is NULL, then the function searches from the first header in the message. Returns NULL when no more header at and after the specified position can be found.

```
pjsip_hdr* pjsip_msg_find_hdr_by_name( pjsip_msg *msg,
                                         const pj_str_t *name,
                                         pjsip_hdr *start);
```

Find header in the *msg* which has the specified *name*, searching both long and short name version of the header from the specified *start* position in the header list. If *start* is NULL, then the function searches from the first header in the message. Returns NULL when no more headers at and after the specified position can be found.

```
void pjsip_msg_add_hdr(    pjsip_msg *msg,
                           pjsip_hdr *hdr);
```

Add *hdr* as the last header in the *msg*.

```
void pjsip_msg_insert_first_hdr( pjsip_msg *msg,
                                  pjsip_hdr *hdr);
```

Add *hdr* as the first header in the *msg*.

```
pj_ssize_t pjsip_msg_print(    pjsip_msg *msg,
                                char *buf,
                                pj_size_t size );
```

Print the whole contents of *msg* to the specified buffer. The function returns the number of bytes written, or -1 if buffer is overflow.

## 2.6 SIP Status Codes

SIP status codes that are defined by the core SIP specification (RFC 3261) is represented by `pjsip_status_code` enumeration in `<pjsip/sip_msg.h>`. In addition, the default reason text can be obtained by calling `pjsip_get_status_text()` function.

The following snippet shows the declaration of the status code in PJSIP.

```
enum pjsip_status_code
{
    PJSIP_SC_TRYING = 100,
    PJSIP_SC_RINGING = 180,
    PJSIP_SC_CALL_BEING_FORWARDED = 181,
    PJSIP_SC_QUEUED = 182,
    PJSIP_SC_PROGRESS = 183,

    PJSIP_SC_OK = 200,

    PJSIP_SC_MULTIPLE_CHOICES = 300,
    PJSIP_SC_MOVED_PERMANENTLY = 301,
    PJSIP_SC_MOVED_TEMPORARILY = 302,
    PJSIP_SC_USE_PROXY = 305,
    PJSIP_SC_ALTERNATIVE_SERVICE = 380,

    PJSIP_SC_BAD_REQUEST = 400,
    PJSIP_SC_UNAUTHORIZED = 401,
    PJSIP_SC_PAYMENT_REQUIRED = 402,
    PJSIP_SC_FORBIDDEN = 403,
    PJSIP_SC_NOT_FOUND = 404,
    PJSIP_SC_METHOD_NOT_ALLOWED = 405,
```

```

PJSIP_SC_NOT_ACCEPTABLE = 406,
PJSIP_SC_PROXY_AUTHENTICATION_REQUIRED = 407,
PJSIP_SC_REQUEST_TIMEOUT = 408,
PJSIP_SC_GONE = 410,
PJSIP_SC_REQUEST_ENTITY_TOO_LARGE = 413,
PJSIP_SC_REQUEST_URI_TOO_LONG = 414,
PJSIP_SC_UNSUPPORTED_MEDIA_TYPE = 415,
PJSIP_SC_UNSUPPORTED_URI_SCHEME = 416,
PJSIP_SC_BAD_EXTENSION = 420,
PJSIP_SC_EXTENSION_REQUIRED = 421,
PJSIP_SC_INTERVAL_TOO_BRIEF = 423,
PJSIP_SC_TEMPORARILY_UNAVAILABLE = 480,
PJSIP_SC_CALL_TSX_DOES_NOT_EXIST = 481,
PJSIP_SC_LOOP_DETECTED = 482,
PJSIP_SC_TOO_MANY_HOPS = 483,
PJSIP_SC_ADDRESS_INCOMPLETE = 484,
PJSIP_SC_AMBIGUOUS = 485,
PJSIP_SC_BUSY_HERE = 486,
PJSIP_SC_REQUEST_TERMINATED = 487,
PJSIP_SC_NOT_ACCEPTABLE_HERE = 488,
PJSIP_SC_REQUEST_PENDING = 491,
PJSIP_SC_UNDECIPHERABLE = 493,

PJSIP_SC_INTERNAL_SERVER_ERROR = 500,
PJSIP_SC_NOT_IMPLEMENTED = 501,
PJSIP_SC_BAD_GATEWAY = 502,
PJSIP_SC_SERVICE_UNAVAILABLE = 503,
PJSIP_SC_SERVER_TIMEOUT = 504,
PJSIP_SC_VERSION_NOT_SUPPORTED = 505,
PJSIP_SC_MESSAGE_TOO_LARGE = 513,

PJSIP_SC_BUSY_EVERYWHERE = 600,
PJSIP_SC_DECLINE = 603,
PJSIP_SC_DOES_NOT_EXIST_ANYWHERE = 604,
PJSIP_SC_NOT_ACCEPTABLE_ANYWHERE = 606,

PJSIP_SC_TSX_TIMEOUT = 701,
PJSIP_SC_TSX_RESOLVE_ERROR = 702,
PJSIP_SC_TSX_TRANSPORT_ERROR = 703,
};

/**
 * Get the default status text for the status code.
 * @param status_code      SIP Status Code
 * @return                  textual message for the status code.
 */
PJ_DECL(const pj_str_t*) pjsip_get_status_text(int status_code);

```

**Code 19 SIP Status Code Constants**

PJSIP also defines new status class (i.e. 7xx) for additional error status during message processing (e.g. transport error, DNS error, etc). This class however is only used internally; it will not go out on the wire.

## 2.7 Non-Standard Parameter Elements

In PJSIP, known or “standard” parameters (e.g. URI parameters, header field parameters) will normally be represented as individual attributes/fields of the corresponding structure. Parameters that are not “standard” will be put in a list of parameters, with each parameter is represented as `pjsip_param` structure. Non-standard parameter normally is declared as `other_param` field in the owning structure.

### 2.7.1 Data Structure Representation (pjsip\_param)

This structure describes each individual parameter in a list.

```
struct pjsip_param
{
    PJ_DECL_LIST_MEMBER(struct pjsip_param); // Generic list member.
    pj_str_t    name;                        // Param/header name.
    pj_str_t    value;                      // Param/header value.
};
```

**Code 20 Non-Standard Parameter Declaration**

For example of its usage, please see `other_param` and `header_param` fields in the declaration of `pjsip_sip_uri` (see previous section 2.1.4 "SIP and SIPS URI") or `other_param` field in the declaration of `pjsip_tel_uri` (see previous section 2.1.5 "Tel URI").

### 2.7.2 Non-Standard Parameter Manipulation

Some functions are provided to assist manipulation of non-standard parameters in parameter list.



```
pjsip_param* pjsip_param_find(    const pjsip_param *param_list,
                                const pj_str_t *name );
```

This function will perform case-insensitive search for the specified parameter name.

```
void pjsip_param_clone(pj_pool_t *pool,
                      pjsip_param *dst_list,
                      const pjsip_param *src_list);
```

Perform full/deep clone of parameter list.

```
void pjsip_param_shallow_clone(  pj_pool_t *pool,
                                pjsip_param *dst_list,
                                const pjsip_param *src_list);
```

Perform shallow clone of parameter list.

```
pj_ssize_t pjsip_param_print_on( const pjsip_param *param_list,
                                char *buf,
                                pj_size_t max_size,
                                const pj_cis_t *pname_unres,
                                const pj_cis_t *pvalue_unres,
                                int sep);
```

Print the parameter list to the specified buffer. The `pname_unres` and `pvalue_unres` is the specification of which characters are allowed to appear unescaped in `pname` and `pvalue` respectively; any characters outside these specifications will be escaped by the function. The argument `sep` specifies separator character to be used between parameters (normally it is semicolon (;) character for normal parameter or comma (,) when the parameter list is a header parameter).

## 2.8 Escapement Rules

PJSIP provides automatic un-escapement during parsing and escapement during printing ONLY for the following message elements:

- all types of URI and their elements are automatically escaped and un-escaped according to their individual escapement rule.
- parameters appearing in all message elements (e.g. in URL, in header fields, etc.) are automatically escaped and un-escaped.

Other message elements will be passed un-interpreted by the stack.

## Chapter 3:Parser

### 3.1 Features

Some features of the PJSIP parser:

- It's a top-down, handwritten parser. It uses PJLIB's scanner, which is pretty fast and reduces the complexity of the parser, which make the parser readable.
- As said above, it's pretty fast. On a single P4/2.6GHz machine, it's able to parse more than 68K of typical 800 bytes SIP message or 860K of 80 bytes URLs in one second. Note that your mileage may vary, and different PJSIP versions may have different performance.
- It's reentrant, which will make it scalable on machine with multi-processors.
- It's extensible. Modules can plug-in new types of header or URI to the parser.

The parser features almost a lot of tricks thinkable to achieve the highest performance, such as:

- it uses zero-copy for all message elements; i.e., when an element, e.g. a *pvalue*, is parsed, the parser does not copy the *pvalue* contents to the appropriate field in the message; instead it will just put the pointer and length to the appropriate field in the message. This is only possible because PJSIP uses *pj\_str\_t* all the way throughout the library, which does not require strings to be NULL terminated.
- it uses PJLIB's memory pool (*pj\_pool\_t*) for memory allocation for the message structures, which provides multiple times speed-up over traditional *malloc()* function.
- it uses zero synchronization. The parser is completely reentrant so that no synchronization function is required.
- it uses PJLIB's try/catch exception framework, which not only greatly simplifies the parser and make it readable, but also saves tedious error checking in the parsers. With an exception framework, only one exception handler needs to be installed at the top-most function of the parser.

One feature that PJSIP parser doesn't implement is *lazy parsing*, which a lot of people probably brag about its usability. In early stage of the design, we decided **not** to implement lazy parsing, because of the following reasons:

- it complicates things, especially error handling. With lazy parsing, basically all parts of the program must be prepared to handle error condition when parsing failed at later stage when application needs to access a particular message element.
- at the end of the day, we believe that PJSIP parser is very fast anyway that it doesn't need lazy parsing. Although having said that, there will be some switches that can be turned-on in PJSIP parser to ignore parsing of some headers for some type of applications (e.g. proxies, which only needs to inspect few header types).

## 3.2 Functions

The main PJSIP parser is declared in `<pjsip/sip_parser.h>` and defined in `<pjsip/sip_parser.c>`. Other parts of the library may provide other parsing functionalities and extend the parser (e.g. `<pjsip/sip_tel_uri.c>` provides function to parse TEL URI and registers this function to the main parser).

### 3.2.1 Message Parsing

```
pj_status_t pjsip_find_msg(    const char *buf,
                               pj_size_t size,
                               pj_bool_t is_datagram,
                               pj_size_t *msg_size);
```

Checks that an incoming packet in *buf* contains a valid SIP message. When a valid SIP message is detected, the size of the message will be indicated in *msg\_size*. If *is\_datagram* is specified, this function will always return `PJ_SUCCESS`.

Note that the function expects the buffer in *buf* to be NULL terminated.

```
pjsip_msg* pjsip_parse_msg(    pj_pool_t *pool,
                               char *buf, pj_size_t size,
                               pjsip_parser_err_report *err_list);
```

Parse a buffer in *buf* into SIP message. The parser will return the message if at least SIP request/status line has been successfully parsed. Any error encountered during parsing will be reported in *err\_list* if this parameter is not NULL.

Note that the function expects the buffer in *buf* to be NULL terminated.

```
pjsip_msg* pjsip_parse_rdata(  char *buf, pj_size_t size,
                               pjsip_rx_data *rdata );
```

Parse a buffer in *buf* into SIP message. The parser will return the message if at least SIP request/status line has been successfully parsed. In addition, this function updates various pointer to headers in *msg\_info* portion of the *rdata*.

Note that the function expects the buffer in *buf* to be NULL terminated.

### 3.2.2 URI Parsing

```
pjsip_uri* pjsip_parse_uri(    pj_pool_t *pool,
                               char *buf, pj_size_t size,
                               unsigned option);
```

Parse a buffer in *buf* into SIP URI. If `PJSIP_PARSE_URI_AS_NAMEADDR` is specified in the *option*, the function will always "wrap" the URI as name address. If `PJSIP_PARSE_URI_IN_FROM_TO_HDR` is specified in the *option*, the function will not parse the parameters after the URI if the URI is not enclosed in brackets (because they will be treated as header parameters, not URI parameters).

This function is able to parse any types of URI that are recognized by the library, and return the correct instance of the URI depending on the scheme.

Note that the function expects the buffer in *buf* to be NULL terminated.

### 3.2.3 Header Parsing

```
void* pjsip_parse_hdr(  pj_pool_t *pool, const pj_str_t *hname,
                        char *line, pj_size_t size,
                        int *parsed_len);
```

Parse the content of a header in *line* (i.e. part of header after the colon character) according to the header type *hname*. It returns the appropriate instance of the header.

Note that the function expects the buffer in *buf* to be NULL terminated.

```

pj_status_t pjsip_parse_headers( pj_pool_t *pool,
                                char *input, pj_size_t size,
                                pj_list *hdr_list );

```

Parse multiple headers found in *input* buffer and put the results in *hdr\_list*. The function expects the header to be separated either by a newline (as in SIP message) or ampersand character (as in URI). The separator is optional for the last header.

Note that the function expects the buffer in *buf* to be NULL terminated.

### 3.3 Extending Parser

The parser can be extended by registering function pointers to parse new types of headers or new types of URI.

```

typedef pjsip_hdr* (pjsip_parse_hdr_func)(pjsip_parse_ctx *context);
pj_status_t pjsip_register_hdr_parser( const char *hname,
                                       const char *hshortname,
                                       pjsip_parse_hdr_func *fptr);

```

Register new function to parse new type of SIP message header.

```

typedef void* (pjsip_parse_uri_func)(pj_scanner *scanner, pj_pool_t *pool,
                                     pj_bool_t parse_params);
pj_status_t pjsip_register_uri_parser( char *scheme,
                                       pjsip_parse_uri_func *func);

```

Register new function to parse new type of SIP URI scheme.



## Chapter 4: Message Buffers

### 4.1 Receive Data Buffer

A SIP message received by PJSIP will be passed around to different PJSIP software components as `pjsip_rx_data` instead of a plain message. This structure contains all information describing the received message.

Receive and transmit data buffers are declared in `<pjsip/sip_transport.h>`.

#### 4.1.1 Receive Data Buffer Structure

```
struct pjsip_rx_data
{
    // This part contains static info about the buffer.
    struct
    {
        pj_pool_t            *pool;           // Pool owned by this buffer
        pjsip_transport      *transport;     // The transport that received the msg.
        pjsip_rx_data_op_key op_key;         // Ioqueue's operation key
    } tp_info;

    // This part contains information about the packet
    struct
    {
        pj_time_val          timestamp;       // Packet arrival time
        char                  packet[PJSIP_MAX_PKT_LEN]; // The packet buffer
        pj_uint32_t          zero;           // Zero padding.
        int                   len;           // Packet length
        pj_sockaddr          addr;           // Source address
        int                   addr_len;      // Address length.
    } pkt_info;

    // This part describes the message and message elements after parsing.
    struct
    {
        char                  *msg_buf;      // Pointer to start of msg in the buf.
        int                   len;           // Message length.
        pjsip_msg             *msg;         // The parsed message.

        // Shortcut to important headers:

        pj_str_t              call_id;      // Call-ID string.
        pjsip_from_hdr        *from;        // From header.
        pjsip_to_hdr          *to;          // To header.
        pjsip_via_hdr         *via;         // First Via header.
        pjsip_cseq_hdr        *cseq;        // CSeq header.
        pjsip_max_forwards_hdr *max_fwd;    // Max-Forwards header.
        pjsip_route_hdr       *route;       // First Route header.
        pjsip_rr_hdr          *record_route; // First Record-Route header.
        pjsip_ctype_hdr        *ctype;      // Content-Type header.
        pjsip_clen_hdr        *clen;        // Content-Length header.
        pjsip_require_hdr     *require;     // The first Require header.

        pjsip_parser_err_report parse_err;  // List of parser errors.
    } msg_info;

    // This part is updated after the rx_data reaches endpoint.
    struct
    {
        pj_str_t              key;           // Transaction key.
        void                  *mod_data[PJSIP_MAX_MODULE]; // Module specific data.
    } endpt_info;
};
```



Code 21 Receive Data Buffer Declaration

## 4.2 Transmit Data Buffer (pjsip\_tx\_data)

When PJSIP application wants to send outgoing message, it must create a transmit data buffer. The transmit data buffer provides memory pool from which all message fields pertaining for the message must be allocated from, a reference counter, lock protection, and other information that are needed by the transport layer to process the message.

```

struct pjsip_tx_data
{
    /** This is for transmission queue; it's managed by transports. */
    PJ_DECL_LIST_MEMBER(struct pjsip_tx_data);

    /** Memory pool for this buffer. */
    pj_pool_t          *pool;

    /** A name to identify this buffer. */
    char                obj_name[PJ_MAX_OBJ_NAME];

    /** For response message, this contains the reference to timestamp when
     * the original request message was received. The value of this field
     * is set when application creates response message to a request by
     * calling #pjsip_endpt_create_response.
     */
    pj_time_val        rx_timestamp;

    /** The transport manager for this buffer. */
    pjsip_tpmgr        *mgr;

    /** Ioqueue asynchronous operation key. */
    pjsip_tx_data_op_key op_key;

    /** Lock object. */
    pj_lock_t          *lock;

    /** The message in this buffer. */
    pjsip_msg          *msg;

    /** Buffer to the printed text representation of the message. When the
     * content of this buffer is set, then the transport will send the content
     * of this buffer instead of re-printing the message structure. If the
     * message structure has changed, then application must invalidate this
     * buffer by calling #pjsip_tx_data_invalidate_msg.
     */
    pjsip_buffer        buf;

    /** Reference counter. */
    pj_atomic_t         ref_cnt;

    /** Being processed by transport? */
    int                 is_pending;

    /** Transport manager internal. */
    void                *token;
    void                (*cb)(void*, pjsip_tx_data*, pj_ssize_t);
};

```

**Code 22 Transmit Data Buffer Declaration**


## Chapter 5: Transports

### 5.1 Transport Class Diagram

TODO.

### 5.2 Using Transports

#### 5.2.1 Function Reference


 `pj_status_t
 pjsip_endpt_acquire_transport(
 pjsip_endpoint *endpt,
 pjsip_transport_type_e t_type,
 const pj_sockaddr_t *remote_addr,
 int addrlen,
 pjsip_transport **p_transport,
 void (*cb)(pj_status_t, pjsip_transport*));`

Acquire transport of type *t\_type* to be used to send message to destination *remote\_addr*. If the function completes immediately and transport is ready to be used, the function will return PJ\_SUCCESS and the transport is returned in *p\_transport* argument. If the function completes immediately with some error, a non-zero error code will be returned. In both cases, the callback will not be called.

If the function can not complete immediately (e.g. TCP is connecting), the function will return PJ\_EPENDING and the callback *cb* will be called when the transport creation process has completed. The status of the pending transport creation is specified in the callback argument.

`pj_status_t pjsip_transport_add_ref( pjsip_transport *transport );`   
 Add reference counter of the *transport*. This function will prevent the transport from being destroyed, and it also cancels idle timer if such timer is active.

`pj_status_t pjsip_transport_dec_ref( pjsip_transport *transport );`   
 Decrement reference counter of the *transport*. When transport's reference counter reaches zero, an idle timer will be started and transport will be destroyed by transport manager when the timer has elapsed and reference counter is still zero.

 `pj_status_t pjsip_transport_send( pjsip_transport *transport,
 pjsip_tx_data *tdata,
 const pj_sockaddr_t *remote_addr,
 int addrlen,
 void *token,
 void (*cb)(void *token,
 pjsip_tx_data *tdata,
 pj_ssize_t bytes_sent));`

Send the message in *tdata* to *remote\_addr* using transport *transport*. If the function completes immediately and data has been sent, the function returns PJ\_SUCCESS. If the function completes immediately with error, a non-zero error code will be returned. In both cases, the callback will not be called.

If the function can not complete immediately (e.g. when the underlying socket buffer is full), the function will return PJ\_EPENDING, and caller will be notified about the completion via the callback *cb*. If the pending send operation completes with error, the error code will be indicated as

negative value of the error code, in the *bytes\_sent* argument of the callback (to get the error code, use "pj\_status\_t status = -bytes\_sent").

## 5.3 Extending Transports

PJSIP transport can be extended to use custom defined transports. Theoretically any types of transport, not limited to TCP/IP, can be plugged into the transport manager's framework. Please see the header file **<pjsip/sip\_transport.h>** and also **sip\_transport\_udp.[hc]** for more details.

## 5.4 Initializing Transports

PJSIP doesn't start any transports by default (not even the built-in transports); it is the responsibility of the application to initialize and start any transports that it wishes to use.

Below are the initialization functions for the built-in UDP and TCP transports.

### 5.4.1 UDP Transport Initialization

PJSIP provides two choices to initialize and start UDP transports. These functions are declared in **<pjsip/sip\_transport\_udp.h>**.

```

pj_status_t pjsip_udp_transport_start(    pjsip_endpoint *endpt,
                                           const pj_sockaddr_in *local_addr,
                                           const pj_sockaddr_in *pub_addr,
                                           unsigned async_cnt,
                                           pjsip_transport **p_transport );

```

Create, initialize, register, and start a new UDP transport. The UDP socket will be bound to *local\_addr*. If the endpoint is located behind firewall/NAT or other port-forwarding devices, then *pub\_addr* can be used as the address that is advertised for this transport; otherwise *pub\_addr* should be the same as *local\_addr*. The argument *async\_cnt* specifies how many simultaneous operations are allowed for this transport, and for maximum performance, the value should be equal to the number of processors in the node.

If transport is successfully started, the function returns PJ\_SUCCESS and the transport is returned in *p\_transport* argument, should the application want to use the transport immediately. Application doesn't need to register the transport to transport manager; this function has done that when the function returns successfully.

Upon error, the function returns a non-zero error code.

```

pj_status_t pjsip_udp_transport_attach(    pjsip_endpoint *endpt,
                                           pj_sock_t sock,
                                           const pj_sockaddr_in *pub_addr,
                                           unsigned async_cnt,
                                           pjsip_transport **p_transport);

```

Use this function to create, initialize, register, and start a new UDP transport when the UDP socket is already available. This is useful for example when application has just resolved the public address of the socket with STUN, and instead of closing the socket and re-create it, the application can just reuse the same socket for the SIP transport.

### **5.4.2 TCP Transport Initialization**

TODO.

### **5.4.3 TLS Transport Initialization**

TODO.

### **5.4.4 SCTP Transport Initialization**

TODO.

## Chapter 6: Stateless Operations

### 6.1 Sending Stateless Response

#### 6.1.1 Base Functions



```

pj_status_t pjsip_endpt_create_response(    pjsip_endpoint *endpt,
                                             pjsip_rx_data *rdata,
                                             int st_code,
                                             const char *st_text,
                                             pjsip_tx_data **tdata);

```

Create a standard response message for the request in *rdata* with status code *st\_code* and status text *st\_text*. If *st\_text* is NULL, default status text will be used.



```

pj_status_t pjsip_endpt_send_response(    pjsip_endpoint *endpt,
                                           pjsip_tx_data *tdata,
                                           pjsip_transport **transport);

```

Send response in *tdata* statelessly. The function will take care of which response destination and transport to use based on the information in the Via header (such as the presence of rport, symmetric transport, etc.)

This function will create a new ephemeral transport if no existing transports can be used to send the message to the destination. The ephemeral transport will be destroyed after some period if it is not used to send any more messages.

If *transport* is not NULL and it points to pointer to transport that is not NULL, that transport will be used. Otherwise if the value is not NULL, it will be filled with the pointer to transport that was used to send the response. This is useful if application wants to prevent the transport from being deleted after some idle period (i.e. by incrementing the reference counter).

The function returns PJ\_SUCCESS if response has been successfully created and send to transport layer, or a non-zero error code. However, even when it returns PJ\_SUCCESS, there is no guarantee that the response has been successfully sent.

#### 6.1.2 Composite Functions



```

pj_status_t pjsip_endpt_respond_stateless( pjsip_endpoint *endpt,
                                             pjsip_rx_data *rdata,
                                             int st_code,
                                             const char *st_text,
                                             const pjsip_hdr *hdr_list,
                                             const pjsip_msg_body *body,
                                             pjsip_transport **transport);

```

This function creates and sends a response to an incoming request. In addition, caller may specify message body and additional headers to be put in the response message in the *hdr\_list* and *body* argument. If there is no additional header or body, to be sent, the arguments should be NULL.

The transport that is used to send the response is returned in *transport* argument if this parameter is not NULL and if transport is immediately available. When transport is not immediately available, the transport *argument* will be set to NULL and the function will send the message only after transport has become available.

The function returns PJ\_SUCCESS if response has been successfully created and send to transport layer, or a non-zero error code. However, even when it returns PJ\_SUCCESS, there is no guarantee that the response has been successfully sent.

### 6.1.3 Usage Examples

#### Sending Account Not Found Response Statelessly

```
static pj_bool_t on_rx_request(pjsip_rx_data *rdata )
{
    pjsip_account *acc;
    pj_status_t status;

    // Find account referred to in the request.
    acc = ...

    // Respond statelessly if account can not be found.
    if (!acc) {
        status = pjsip_endpt_respond_stateless( endpt, rdata, 404, NULL /*Not Found*/,
                                                NULL, NULL, NULL);

        return PJ_TRUE;
    }

    // Process the account
    ...

    return PJ_TRUE;
}
```

Code 23 Sample: Stateless Response

#### Handling Authentication Failures Statelessly

Another (longer) way to send stateless response:

```
static pj_bool_t on_rx_request( pjsip_rx_data *rdata )
{
    pjsip_account *acc;

    // Lookup acc.
    acc = ...;

    // Check authorization and handle failure statelessly
    if (!pjsip_auth_authorize( acc, rdata->msg )) {
        pjsip_proxy_authenticate_hdr *auth_hdr;

        status = pjsip_endpt_create_response( endpt, rdata,
                                              407, NULL /* Proxy Auth Required */,
                                              &tdata);

        // Add Proxy-Authenticate header.
        status = pjsip_auth_create_challenge( tdata->pool, ..., &auth_hdr);
        pjsip_msg_add_hdr( &tdata->msg, auth_hdr );

        // Send response statelessly
        status = pjsip_endpt_send_response( endpt, tdata, NULL);
        return PJ_TRUE;
    }

    // Authorization success. Proceed to next stage..
    ...
    return PJ_TRUE;
}
```

Code 24 Sample: Stateless Response

#### Stateless Redirection

```

static pj_bool_t on_rx_request( pjsip_rx_data *rdata )
{
    pjsip_account *acc;
    pj_status_t status;

    // Find the account referred to in the request.
    acc = ...
    if (!acc) {
        status = pjsip_endpt_respond_stateless( endpt, rdata, 404, NULL /*Not Found*/,
                                                NULL, NULL, NULL );
        return PJ_TRUE;
    }

    //
    // Send 301/Redirect message, specifying the Contact details in the response
    //
    status = pjsip_endpt_respond_stateless( endpt, rdata,
                                            301, NULL /*Moved Temporarily*/,
                                            &acc->contact_list, NULL, NULL);

    return PJ_TRUE;
}

```

**Code 25 Stateless Redirection**

## 6.2 Sending Stateless Request

### 6.2.1 Related Functions

```

pj_status_t pjsip_endpt_create_tdata( pjsip_endpoint *endpt,
                                     pjsip_tx_data **tdata);

```

Create a new, blank transmit data.



```

pj_status_t pjsip_endpt_send_request_stateless( pjsip_endpoint *endpt,
                                                pjsip_tx_data *tdata,
                                                pjsip_transport **transport);

```

Send request in *tdata* statelessly. The function will take care of which destination and transport to use based on the information in the message, taking care of URI in the request line and Route header.

If *transport* is specified, that transport will be used. Otherwise if the value is not NULL and transport was immediately available, the function sets the value of *transport* parameter with transport that was used to send the response.

### 6.2.2 Usage Examples

#### Sending Request Statelessly

```

void my_send_request()
{
    pj_status_t status;
    pjsip_tx_data *tdata;

    // Create the request.
    // Actually the function takes pj_str_t* argument instead of char*.
    status = pjsip_endpt_create_request( endpt,                // endpoint
                                        method,                // method
                                        "sip:bob@example.com",   // target URI
                                        "sip:alice@thishost.com", // From:
                                        "sip:bob@example.com",   // To:
                                        "sip:alice@thishost.com", // Contact:
                                        NULL,                    // Call-Id
                                        0,                        // CSeq#

```



```

NULL, // body
&tdata ); // output

// You may modify the message before sending it.
...

// Send the request statelessly.
status = pjsip_endpt_send_request_stateless( endpt, tdata, NULL);
}

```

**Code 26 Sending Stateless Request**

## 6.3 Stateless Proxy Forwarding

Proxy may choose to forward a request statelessly. When doing so however, it must strictly follow guidelines in section **16.11 Stateless Proxy** of RFC 3261.

### 6.3.1 Base Functions



```

pj_status_t pjsip_endpt_create_request_fwd(pjsip_endpoint *endpt,
                                           pjsip_rx_data *rdata,
                                           const pjsip_uri *uri,
                                           const pj_str_t *branch,
                                           unsigned options,
                                           pjsip_tx_data **tdata);

```

Create new request message to be forwarded upstream to new destination URI *uri*. The new request is a full/deep clone of the request received in *rdata*, unless if other copy mechanism is specified in the *options*. The *branch* parameter, if not NULL, will be used as the branch-param in the Via header. If it is NULL, then a unique branch parameter will be used.



```

pj_status_t pjsip_endpt_create_response_fwd( pjsip_endpoint *endpt,
                                              pjsip_rx_data *rdata,
                                              unsigned options,
                                              pjsip_tx_data **tdata);

```

Create new response message to be forwarded downstream by the proxy from the response message found in *rdata*. Note that this function practically will clone the response as is, i.e. without checking the validity of the response or removing top most Via header. This function will perform full/deep clone of the response, unless other copy mechanism is used in the *options*.



```

pj_str_t pjsip_calculate_branch_id( pjsip_rx_data *rdata );

```

Create a globally unique branch parameter based on the information in the incoming request message. This function guarantees that subsequent retransmissions of the same request will generate the same branch id.

This function can also be used in the loop detection process. If the same request arrives back in the proxy with the same URL, it will calculate into the same branch id.

Note that the returned string was allocated from *rdata*'s pool.

## 6.3.2 Usage Examples

### Stateless Forwarding

```
static pj_bool_t on_rx_request( pjsip_rx_data *rdata )
{
    pjsip_account *acc;
    pjsip_tx_data *tdata;
    pj_str_t branch_id;
    pj_status_t status;

    // Find the account specified in the request.
    acc = ...

    // Generate unique branch ID for the request.
    branch_id = pjsip_calculate_branch_id( rdata );

    // Create new request to be forwarded to new destination.
    status = pjsip_endpt_create_request_fwd( endpt, rdata, dest, &branch_id, 0,
                                             &tdata );

    // The new request is good to send, but you may modify it if necessary
    // (e.g. adding/replacing/removing headers, etc.)
    ...

    // Send the request downstream
    status = pjsip_endpt_send_request_stateless( endpt, tdata, NULL );

    return PJ_TRUE;
}

//
// Forward response upstream
//
static pj_bool_t on_rx_response( pjsip_rx_data *rdata )
{
    pjsip_tx_data *tdata;
    pj_status_t status;

    // Check that topmost Via is ours, strip top-most Via, etc.
    ...

    // Create new tdata for the response.
    status = pjsip_endpt_create_response_fwd( endpt, rdata, 0, &tdata );

    // Send the response upstream
    status = pjsip_endpt_send_response( endpt, tdata, NULL );

    return PJ_TRUE;
}
```

**Code 27 Stateless Forwarding**

## Chapter 7: Transactions

### 7.1 Design

#### 7.1.1 Introduction

Transaction in PJSIP is represented with `pjsip_transaction` structure in header file `<pjsip/sip_transaction.h>`. Transaction's lifetime normally follows these steps:

- Created by `pjsip_endpt_create_uac_tsx()` / `pjsip_endpt_create_uas_tsx()`.
- When application wants to send a message using the transaction, it will call `pjsip_tsx_send_msg()`.
- Transaction state automatically changes as messages are passed to it (either by endpoint for incoming message or by transaction user for outgoing message) or timer elapses, and transaction user is notified via the callback.
- Transaction will be automatically destroyed once it the state has reached `PJSIP_TSX_STATE_TERMINATED`. Application can also forcibly terminate the transaction by calling `pjsip_tsx_terminate()`.

#### 7.1.2 Timers and Retransmissions

Transaction only has two types of timers: retransmission timer and timeout timer. The value of both timer types are automatically set by the transaction according to the transaction type (UAS or UAC), transport (reliable or non-reliable), and method (INVITE or non-INVITE).

Application can change the interval value of timers only on a global basis (perhaps even only during compilation).

A transaction handles both incoming and outgoing retransmissions. Incoming retransmissions are silently absorbed and ignored by transaction; there is no notification about incoming retransmissions emitted by transaction. Outgoing messages are automatically retransmitted by transactions where necessary; again there will be no notification emitted by transaction on outgoing retransmissions.

#### 7.1.3 INVITE Final Response and ACK Request

##### Failed INVITE Request



The transaction behaves **exactly** according to RFC 3261 for failed INVITE request.

**Client transaction:** when a client INVITE transaction receives 300-699 final response to INVITE, it will automatically emit ACK request to the response. The transaction then wait for timer D interval before it is terminated, during which any incoming 300-699 response retransmissions will be automatically answered with ACK request.

**Server transaction:** when a server INVITE transaction is asked to transmit 300-699 final response, it will transmit the response and keep retransmitting the response until an ACK request is received or timer H interval has elapsed. During this interval, when ACK request is received, transaction will move to Confirmed state and will be destroyed after timer I interval has elapsed. When timer H elapsed without receiving a valid ACK request, transaction will be destroyed.

### Successful INVITE Request

**Client transaction:** when a client INVITE transaction receives 2xx final response to INVITE, it will destroy itself automatically after it passes the response to its transaction user (can be a dialog or application). Subsequent incoming 2xx response retransmission will be passed directly to dialog or application.

In any case, application **MUST** send ACK request manually upon receiving 2xx final response to INVITE.

**Server transaction:** when a server INVITE transaction is asked to transmit 2xx final response, it will transmit the response and **keep retransmitting** the response until ACK is received or transaction is terminated by application with `pjsip_tsx_terminate()`.

For simplicity in the implementation, a typical UAS dialog normally will let the transaction handle the retransmission of the 2xx INVITE response. But proxy application **MUST** destroy the UAS transaction as soon as it receives and sends the 2xx response, to allow the 2xx retransmission to be handled by end-to-end user agents.



This behavior of INVITE server transaction is **different** than RFC 3261 for successful INVITE request, which says that INVITE server transaction **MUST** be destroyed once 2xx response is sent. The PJSIP transaction behavior allows more simplicity in the dialog implementation, while maintaining the flexibility to be compliant with RFC 3261 for proxy applications.

### 7.1.4 Incoming ACK Request

When the server INVITE transaction was completed with non-successful final response, the ACK request will be absorbed by transaction; transaction user **WILL NOT be notified** about the incoming ACK request.

When the server INVITE transaction was completed with 2xx final response, the first ACK request will be notified to transaction user. Subsequent receipt of ACK retransmission **WILL NOT** be notified to transaction user.

### 7.1.5 Server Resolution and Transports

Transaction will handle server and transport resolution automatically. Server resolution is performed according to RFC 3263 (Locating SIP Servers) and is done asynchronously using PJSIP's resolver framework. Once the server has been located, the transaction will request transport framework to establish transport connection to the server.

The transaction adds reference counter to the transport it currently uses.

The transaction also automatically handles fail-over to another server whenever it fails to contact the server, provided that alternative server is available (as returned by server resolution framework). Similarly it will try to use other transport when current transport fails to deliver the message, according to rules specified in RFC 3261 chapter 18 "Transports".



**Note:** the transaction **DOES NOT** maintain timeout value while resolving server or connecting transport. Thus it is possible that the total duration of a transaction exceeds the transaction timeout specification (timer B and F) because of a lengthy server resolution or transport connection establishment operation.

For server resolution process, the transaction sets the resolution timeout to timer B or F; if server can not be resolved during this time, the transaction will fail. For TCP/TLS transport establishment,

transaction DOES NOT set any timeout value; it relies on TCP timeout to detect connection failures.

### TCP Connection Closure

A TCP connection closure will not automatically cause the transaction to fail. In fact, the transaction will not even detect the failure until it tries to send a message. When it does, it follows the normal procedure to send the message using alternative transport.

## 7.1.6 Via Header

### Branch Parameter

If Via header is not present in the request, a new one will be created along with a globally unique branch ID. If both Via header and branch parameter are present, transaction will use the branch parameter value as the transaction identifier. If branch parameter is not present, a unique branch ID will be created and used.

### Via Sent-By

Transaction automatically puts the appropriate transport information in the sent-by part of the Via header. This allows the transaction to switch to another transport when current transport fails to deliver the message.

## 7.2 Reference

### 7.2.1 Base Functions



```

pj_status_t pjsip_endpt_create_uac_tsx( pjsip_endpoint *endpt,
                                         pjsip_module *tsx_user,
                                         pjsip_tx_data *tdata,
                                         pjsip_transaction **p_tsx );

```

Create a new UAC transaction for the outgoing request in *tdata* with the transaction user set to *tsx\_user*. The transaction is automatically initialized and registered to endpoint's transaction table. Note that after calling this function, applications normally would call `pjsip_tsx_send_msg()` to actually send the request.



```

pj_status_t pjsip_endpt_create_uas_tsx( pjsip_endpoint *endpt,
                                         pjsip_module *tsx_user,
                                         pjsip_rx_data *rdata,
                                         pjsip_transaction **p_tsx );

```

Create a new UAS transaction for the incoming request in *rdata* with the transaction user set to *tsx\_user*. The transaction is automatically initialized and registered to endpoint's transaction table.



```

pj_status_t pjsip_tsx_send_msg( pjsip_transaction *tsx,
                                pjsip_tx_data *tdata );

```

Send message through the transaction. If *tdata* is NULL, the last message or the message that was specified during creation will be retransmitted.

```

pj_status_t pjsip_tsx_create_key( pj_pool_t *pool,
                                  pj_str_t *out_key,
                                  pjsip_role_e role,
                                  const pjsip_method *method,
                                  const pjsip_rx_data *rdata );

```

Create a transaction key from an incoming request or response message, taking into consideration whether the message is compliant with RFC 3261 or RFC 2543. The key can be used to find the transaction in endpoint's transaction table.

The function returns the key in *out\_key* parameter. The *role* parameter is used to find either UAC or UAS transaction, and the *method* parameter contains the method of the message.



```
pjsip_transaction* pjsip_endpt_find_tsx(    pjsip_endpoint *endpt,
                                           const pj_str_t *key,
                                           pj_bool_t lock );
```

Find transaction with the specified key in endpoint's transaction table. If *lock* parameter is non-zero, this function will also lock the transaction before returning the transaction, so that other threads are not able to delete the transaction. Caller then is responsible to unlock the transaction when it's finished using the transaction.

```
pj_status_t pjsip_tsx_terminate( pjsip_transaction *tsx,
                                int st_code );
```

Forcefully terminate the transaction *tsx* with the specified status code *st\_code*. Normally application doesn't need to call this function, since transactions will terminate and destroy themselves according to their state machine.

This function can be used for example when 200/OK response to INVITE is sent/received and the UA layer wants to handle retransmission of 200/OK response manually.

The transaction will emit transaction state changed event (state changed to `PJSIP_TSX_STATE_TERMINATED`), then it will be unregistered and destroyed immediately by this function.



```
pjsip_rx_data* pjsip_rdata_get_tsx ( pjsip_rx_data *rdata );
```

Get the transaction object in an incoming message.

## 7.2.2 Composite Functions



```
pj_status_t pjsip_endpt_respond( pjsip_endpoint *endpt,
                                pjsip_module *tsx_user,
                                pjsip_rx_data *rdata,
                                int st_code,
                                const char *st_text,
                                const pjsip_hdr *hdr_list,
                                const pjsip_msg_body *body,
                                pjsip_transaction **p_tsx)
```

Send respond by creating a new UAS transaction for the incoming request.

```
pj_status_t pjsip_endpt_send_request(pjsip_endpoint *endpt,
                                    pjsip_tx_data *tdata,
                                    int timeout,
                                    void *token,
                                    void (*cb)(void*, pjsip_event*))
```

Send the request by using an UAC transaction, and optionally request callback to be called when the transaction completes.

## 7.3 Sending Statefull Responses

### 7.3.1 Usage Examples

#### Sending Response Statefully (The Hard Way)

```
static pj_bool_t on_rx_request( pjsip_rx_data *rdata )
{
    pj_status_t status;
    pjsip_transaction *tsx;
    pjsip_tx_data *tdata;
```

```

// Create and initialize transaction.
status = pjsip_endpt_create_uas_tsx( endpt, NULL, rdata, &tsx );

// Create response
status = pjsip_endpt_create_response( endpt, rdata, 200, NULL /*OK*/, &tdata);

// The response message is good to send, but you may modify it before
// sending the response.
...

// Send response with the specified transaction.
pjsip_tsx_send_msg( tsx, tdata );

return PJ_TRUE;
}

```

**Code 28 Sending Statefull Response****Sending Response Statefully (The Easy Way)**

```

static pj_bool_t on_rx_request( pjsip_rx_data *rdata )
{
    pj_status_t status;

    // Respond to the request statefully
    status = pjsip_endpt_respond( endpt, NULL, rdata,
                                   200, NULL /* OK */, NULL, NULL, NULL);

    return PJ_TRUE;
}

```

**Code 29 Sending Statefull Response**

## 7.4 Sending Statefull Request

Two ways to send statefull request:

- use `pjsip_endpt_send_request()`
- using transaction manually.

### 7.4.1 Usage Examples

**Sending Request with Completion Callback**

```

void my_send_request()
{
    pj_status_t status;
    pjsip_tx_data *tdata;

    // Create the request.
    status = pjsip_endpt_create_request( endpt, ..., &tdata );

    // You may modify the message before sending it.
    ...

    // Send the request statefully.
    status = pjsip_endpt_send_request( endpt, tdata, -1, NULL, &on_complete);
}

static void on_complete( void *token, pjsip_event *event )
{
    pj_assert(event->type == PJSIP_EVENT_TSX_STATE);
}

```

```
PJ_LOG(3,("app", "Transaction completes, status_code=%d",
          event->body.tsx_state.tsx->status_code));
}
```

**Code 30 Sending Request Statefully****Sending Request with Transaction**

```
extern pjsip_module app_module;

void my_send_request()
{
    pj_status_t status;
    pjsip_tx_data *tdata;
    pjsip_transaction *tsx;

    // Create the request.
    status = pjsip_endpt_create_request( endpt, ..., &tdata );

    // You may modify the message before sending it.
    ...

    // Create transaction.
    status = pjsip_endpt_create_uac_tsx( endpt, &app_module, tdata, &tsx );

    // Send the request.
    status = pjsip_tsx_send_msg( tsx, tdata /*or NULL*/ );
}

static void on_tsx_state( pjsip_transaction *tsx, pjsip_event *event )
{
    pj_assert(event->type == PJSIP_EVENT_TSX_STATE);
    PJ_LOG(3,("app", "Transaction %s: state changed to %s",
              tsx->obj_name, pjsip_tsx_state_str(tsx->state)));
}
```

**Code 31 Sending Request Statefully**

## 7.5 Statefull Proxy Forwarding

### 7.5.1 Usage Examples

**Statefull Forwarding**

The following code shows a sample statefull forwarding proxy. The code creates UAS and UAC transaction (one for each side), forward the request to the UAC side, and forward all responses from the UAC side to UAS side. It also handles transaction timeout or other error in the UAC side and sends response to the UAS side.

One that it doesn't handle is receiving CANCEL request in the UAC side.

```
// This is our proxy module.
extern pjsip_module proxy_module;

static pj_bool_t on_rx_request( pjsip_rx_data *rdata )
{
    pjsip_account *acc;
    pjsip_uri *dest;
    pjsip_transaction *uas_tsx, *uac_tsx;
    pjsip_tx_data *tdata;
    pj_status_t status;

    // Find the account specified in the request.
    acc = ...

    // Respond statelessly with 404/Not Found if account can not be found.
```



```

if (!acc) {
    ...
    return PJ_TRUE;
}
// Set destination URI from account's contact list that has highest priority.
dest = ...

// Create UAS transaction
status = pjsip_endpt_create_uas_tsx( endpt, &proxy_module, rdata, &uas_tsx);

// Copy request to new tdata with new target URI.
status = pjsip_endpt_create_request_fwd( endpt, rdata, dest, NULL, 0, &tdata);

// Create new UAC transaction.
status = pjsip_endpt_create_uac_tsx( endpt, &proxy_module, tdata, &uac_tsx );

// "Associate" UAS and UAC transaction
uac_tsx->mod_data[proxy_module.id] = (void*)uas_tsx;
uas_tsx->mod_data[proxy_module.id] = (void*)uac_tsx;

// Forward message to UAC side
status = pjsip_tsx_send_msg( uac_tsx, tdata );
return PJ_TRUE;
}

static pj_bool_t on_rx_response( pjsip_rx_data *rdata )
{
    pjsip_transaction *tsx;
    pjsip_tx_data *tdata;
    pj_status_t status;

    // Get transaction object in rdata.
    tsx = pjsip_rdata_get_tsx( rdata );

    // Check that this transaction was created by the proxy
    if (tsx->tsx_user == &proxy_module) {
        // Get the peer UAC transaction.
        pjsip_transaction *uas_tsx;
        uas_tsx = (pjsip_transaction*) tsx->mod_data[proxy_module.id];

        // Check top-most Via is ours
        ...
        // Strip top-most Via
        // Note that after this code, rdata->msg_info.via is invalid.
        pj_list_erase(rdata->msg_info.via);
        // Code above is equal to:
        // pjsip_hdr *via = pjsip_msg_find_hdr(rdata->msg, PJSIP_H_VIA);
        // pj_list_erase(via);

        // Copy the response msg.
        status = pjsip_endpt_create_response_fwd( endpt, rdata, 0, &tdata);

        // Forward the response upstream.
        pjsip_tsx_send_msg( uas_tsx, tdata );

        return PJ_TRUE;
    }
    ...
}

```

**Code 32 Statefull Forwarding**

## Chapter 8: Authentication Framework

PJSIP provides framework for performing both client and server authentication. The authentication framework supports HTTP digest authentication by default, but other authentication schemes may be added to the framework.

The following diagram illustrates the framework's "class diagram".

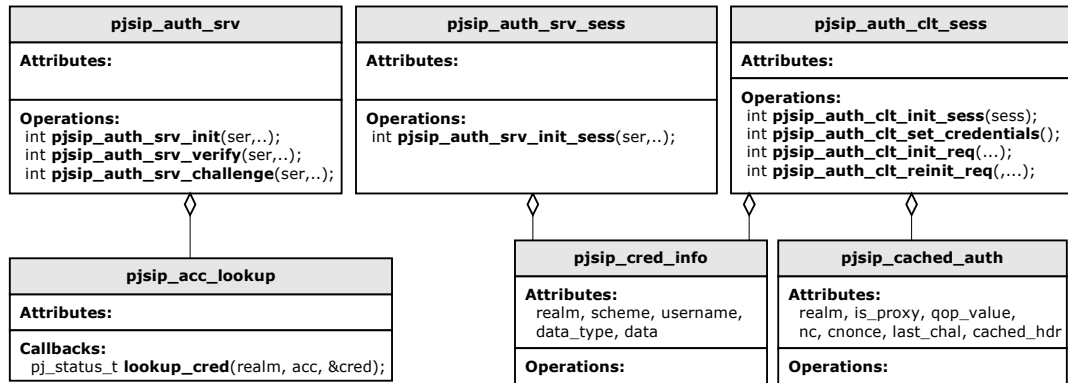


Figure 12 Authentication Framework

### 8.1 Client Authentication Framework

The client authentication framework manages authentication process by client to all downstream servers. It automatically responds to server's challenge with the correct credential (when such credential is supplied), cache the authorization info, and initialize subsequent requests with the cached authorization info.

#### 8.1.1 Client Authentication Framework Reference

##### Data Structure Reference

Structure	Description
<b>pjsip_cred_info</b>	<p>This structure describes the credential to be used to authenticate against a specific realm. A client can have multiple credentials to use for the duration of a dialog or registration; each one of the credential contains information needed to authenticate against a particular downstream proxy or server.</p> <p>For example, client need one credential to authenticate against its outbound proxy, and another credential to authenticate against the end server.</p>
<b>pjsip_cached_auth</b>	<p>This structure keeps the last challenge received from a particular server. It is needed so that client can initialize next request with the last challenge.</p>
<b>pjsip_auth_client_session</b>	<p>This structure describes the client authentication session. Client would normally keep this structure for the duration of a dialog or client registration.</p>

Figure 13 Client Authentication Data Structure

## Function Reference



```
pj_status_t pjsip_auth_client_init( pjsip_auth_client_session *sess,
                                     pj_pool_t *pool, unsigned options);
```

Initialize client authentication session data structure, and set the session to use *pool* for its subsequent memory allocation. The argument *options* should be set to zero for this PJSIP version.



```
pj_status_t pjsip_auth_client_set_credentials( pjsip_auth_client_session *s,
                                                int cred_cnt,
                                                const pjsip_cred_info cred[]);
```

Set the credentials to be used during the session.



```
pj_status_t pjsip_auth_client_init_req( pjsip_auth_client_session *sess,
                                          pjsip_tx_data *tdata );
```

This function add all relevant authorization headers to a new outgoing request *tdata* according to the cached information in the session. The request line in the request message must be valid before calling this function.



```
pj_status_t pjsip_auth_client_reinit_req( pjsip_auth_client_session *sess,
                                            pjsip_endpoint *endpt,
                                            const pjsip_rx_data *rdata,
                                            pjsip_tx_data *old_request,
                                            pjsip_tx_data **new_request );
```

Call this function to re-initialize a request upon receiving failed authentication status (401/407 response). This function will recreate *new\_request* according to *old\_request*, and add appropriate Authorization and Proxy-Authorization headers according to the challenges found in *rdata* response. In addition, this function also put the relevant information in the session.

This function will return failure if there is a missing credential for the challenge. Note that this function may reuse the old request instead of creating a fresh one.

## 8.1.2 Examples

### Client Transaction Authentication

The following example illustrates how to initialize outgoing request with authorization information and how to handle challenge received from the server. For brevity, error handling is not shown in the example. A real application should be prepared to handle error situation in all stages.

```
pjsip_auth_client_session auth_sess;

// Initialize client authentication session with some credentials.
void init_auth(pj_pool_t *session_pool)
{
    pjsip_cred_info cred;
    pj_status_t status;

    cred.realm = pj_str("sip.example.com");
    cred.scheme = pj_str("digest");
    cred.username = pj_str("alice");
    cred.data_type = PJSIP_CRED_DATA_PLAIN_PASSWD;
    cred.data = pj_str("secretpassword");

    status = pjsip_auth_client_init( &auth_sess, session_pool, 0);
    status = pjsip_auth_set_credentials( &auth_sess, 1, &cred );
}

// Initialize outgoing request with authorization information and
// send the request statefully.
```

```

void send_request(pjsip_tx_data *tdata)
{
    pj_status_t status;

    status = pjsip_auth_client_init_req( &auth_sess, tdata );
    status = pjsip_endpt_send_request( endpt, tdata, -1, NULL, &on_complete);
}

// Callback when the transaction completes.
static void on_complete( void *token, pjsip_event *event )
{
    int code;

    pj_assert(event->type == PJSIP_EVENT_TSX_STATE);
    code = event->body.tsx_state.tsx->status_code;
    if (code == 401 || code == 407) {
        pj_status_t status;
        pjsip_tx_data *new_request;

        status = pjsip_auth_client_reinit_req( &auth_sess, endpt,
                                                event->body.tsx_state.src.rdata,
                                                tsx->last_tx,
                                                &new_request);

        if (status == PJ_SUCCESS)
            status = pjsip_endpt_send_request( endpt, new_request, -1, NULL,
                                                &on_complete);
        else
            PJ_LOG(3, ("app", "Authentication failed!!!"));
    }
}

```

**Code 33 Client Athorization Example**

## 8.2 Server Authorization Framework

The server authorization framework provides two types of server authorization mechanisms:

- session-less server authorization, which provides general API for authenticating clients. This API provides global server authorization mechanism on request-per-request basis, and is normally used for proxy application where it doesn't have the notion of dialog.
- server authorization session, which provides API for authenticating requests inside a particular dialog or registration session. One server authorization session instance needs to be created for each server side dialog or registration session. A server auth session will have exactly one credential which is setup initially, and this credential must be used by client throughout the duration of the dialog/registration session.

The server authorization session currently is not implemented. Only global, session-less server authorization framework is available.

### 8.2.1 Server Authorization Reference

#### Data Types Reference



```

typedef pj_status_t pjsip_auth_lookup_cred(pj_pool_t *pool,
                                           const pj_str_t *realm,
                                           const pj_str_t *acc_name,
                                           pjsip_cred_info *cred_info );

```

Type of function to be registered to authorization server to lookup for credential information for the specified *acc\_name* in the specified *realm*. When credential information is successfully retrieved, the function must fill in the *cred\_info* with

the credentials and return PJ\_SUCCESS. Otherwise it should return one of the following error code:

- PJSIP\_EAUTHACCNOTFOUND: account not found for the specified realm,
- PJSIP\_EAUTHACCDISABLED: account was found but disabled,

## Functions Reference



```
pj_status_t pjsip_auth_srv_init( pj_pool_t *pool,
                                pjsip_auth_srv *auth_srv,
                                const pj_str_t *realm,
                                pjsip_auth_lookup_cred *lookup_func,
                                unsigned options );
```

Initialize server authorization session data structure to serve the specified *realm* and to use *lookup\_func* function to look for the credential info. The argument *options* is bitmask combination of the following values:

- PJSIP\_AUTH\_SRV\_IS\_PROXY: to specify that the server will authorize clients as a proxy server (instead of as UAS), which means that Proxy-Authenticate will be used instead of WWW-Authenticate.



```
pj_status_t pjsip_auth_srv_verify( pjsip_auth_srv *auth_srv,
                                   pjsip_rx_data *rdata );
```

Request the authorization server framework to verify the authorization information in the specified request in *rdata*. This function will return PJ\_SUCCESS if the authorization information found in the request can be accepted, or the following error when authorization failed:

- PJSIP\_EAUTHACCNOTFOUND or PJSIP\_EAUTHACCDISABLED are the error codes returned by the lookup function.
- PJSIP\_EAUTHINVALIDREALM: invalid realm,
- PJSIP\_EAUTHINVALIDDIGEST: invalid digest,
- other non-zero values may be returned to indicate system error.



```
pj_status_t pjsip_auth_srv_challenge( pjsip_auth_srv *auth_srv,
                                       const pj_str_t *nonce,
                                       const pj_str_t *opaque,
                                       pj_bool_t stale,
                                       pjsip_tx_data *tdata );
```

Add authentication challenge headers to the outgoing request in *tdata*. Application may specify its customized *nonce* and *opaque* for the challenge, or can leave the value to NULL to make the function fills them in with random characters.

## 8.3 Extending Authentication Framework

The authentication framework can be extended to support authentication framework other than HTTP digest (e.g. PGP, etc.).

TODO.

## Chapter 9: Basic User Agent Layer (UA)

### 9.1 Basic Dialog Concept

The basic UA dialog provides basic facilities for managing SIP dialogs, such as basic dialog state, session counter, common Call-ID, From, To, and Contact headers, sequencing of CSeq in transactions, and common route-set.

The basic UA dialog is agnostic/skeptical of what kind of sessions it is being used to (e.g. INVITE session, SUBSCRIBE/NOTIFY sessions, REFER/NOTIFY sessions, etc.), and it can be used to establish multiple and different types of sessions simultaneously in a single dialog.

A PJSIP dialog can be considered just as a passive data structure to hold common dialog attributes. You must not confuse dialog with an INVITE session. An INVITE session is a session (also commonly known as *dialog usage*) "inside" a dialog. There can be other sessions/usages in the same dialog; all of them share common dialog properties (although there can only be one INVITE session per dialog).



For more information about dialog-usage concept, please refer to **draft-sparks-sipping-dialogusage-01.txt**. The document identifies two dialog-usages, i.e. invite usage and subscribe usage.

PJSIP dialog does not know the state of its sessions. It doesn't know whether the INVITE session has been established or disconnected. In fact, PJSIP dialog does not even know what kind of sessions are there in the dialog. All it cares is how many active sessions are there in the dialog. The dialog is started with one active session, and when the session counter reaches zero and the last transaction is terminated, the dialog will be destroyed.

It will be the responsibility of each dialog usages to increment and decrement the dialog's session counter.

#### 9.1.1 Dialog Sessions

Dialog sessions in PJSIP dialog framework is just represented with a reference counter. This reference counter is incremented and decremented by dialog usage module everytime it creates/destroys a session in that particular dialog.

Dialog's sessions are created by dialog usages. In one particular dialog, one dialog usage can create more than one sessions (except invite usage, which can only create one invite session in a single dialog).

#### 9.1.2 Dialog Usages

Dialog usages are PJSIP modules that are registered to the dialog to receive dialog's events. Multiple modules can be registered to one dialog, hence the dialog can have multiple usages. Each dialog usage module is responsible to handle a specific session. For example, the subscribe usage module will create a new subscribe session each time it receives new SUBSCRIBE request (and increment dialog's session counter), and decrement the session counter when the subscribe session has terminated.

The processing of dialog usages by a dialog is similar to the processing of modules by endpoint; for each `on_rx_request()`, `on_rx_response()`, and `on_tsx_state()` events, the dialog passes the event to each dialog usages starting

from the higher priority module (i.e. the one with lower priority number) until one of the module returns true (i.e. non-zero), which in this case the dialog will stop the distribution of the event further.

In its most basic (i.e. low-level) use, the application manages the dialog directly, and it is the only “usage” (or user) of the dialog. In this case, the application is responsible for managing the sessions inside the dialog, which means handling ALL requests and responses and establishing/tearing down sessions manually.

In later chapters, we will learn about high-level APIs that can be used to manage sessions. These high-level APIs are PJSIP modules that are registered to the dialog as dialog usages, and they will handle/react to different types of SIP messages that are specific to each type of sessions (e.g. an invite usage module will handle INVITE, PRACK, CANCEL, ACK, BYE, UPDATE and INFO, a subscribe usage module will handle REFER, SUBSCRIBE, and NOTIFY, etc.). These high level APIs provide high-level callbacks according to the session’s specification.

In this chapter however, we’ll only lean about basic, low-level dialog usage.

### 9.1.3 Class Diagram

The basic dialog class diagram is as follows.

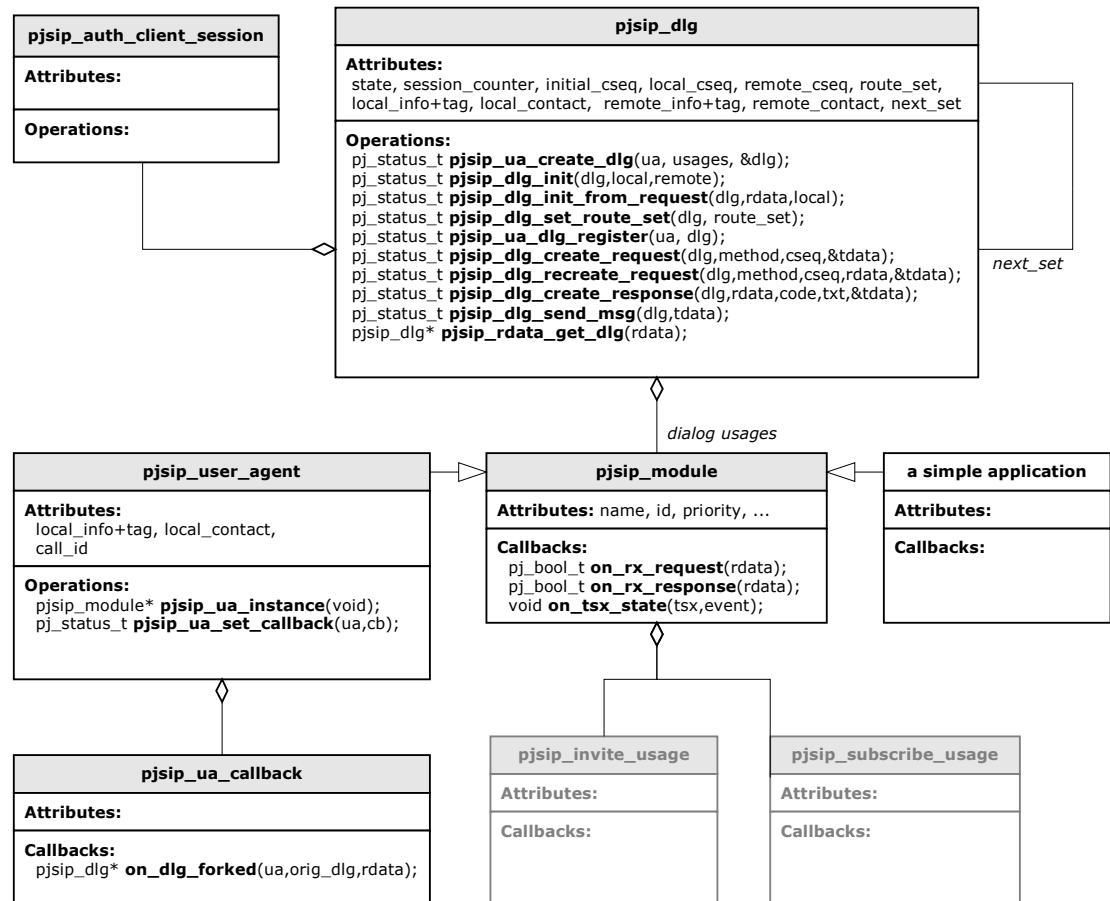


Figure 14 Basic User Agent Class Diagram

The diagram shows the relationship between dialog and its usages. In the most basic/low-level scenario, the application module is the only usage of the dialog. In more high-level scenario, some high-level modules (e.g. **pjsip\_invite\_usage** and **pjsip\_subscribe\_usage**) can be registered to a dialog as dialog’s usages, and the

application will receive events from these usages instead instead of directly from the dialog.

The *next\_set* relationship is the relationship between a dialog and other dialogs in the same dialog set, which was created when the dialog forked.

The diagram also shows PJSIP user agent module (`pjsip_user_agent`). The user agent module is the "owner" of all dialogs; it's the module that distributes events from transaction layer to the corresponding dialog. Although user agent module is derived from `pjsip_module`, note that the user agent module is NOT a dialog usage.

The diagram also shows the relationship between a dialog and client authorization session (`pjsip_auth_client_session`). Each dialog has exactly one client authorization session to handle all authorization challenges for all outgoing requests.

## 9.1.4 Forking

### Handling Forking Condition

The user agent module provides a callback that can be registered by application when it detects forked response from the downstream proxy. A forked response is defined as a response (can be provisional or 2xx response) within a dialog that has To tag that is different from any of existing dialogs. When such responses are received, the user agent will call `on_dlg_forked()` callback, passing the received response and the original dialog (the dialog that application created originally) as the arguments.



**It is the complete responsibility of the application to handle forking condition!**

Upon receiving a forked provisional response, application can:

- ignore the provisional response (perhaps waiting until a final, forked 2xx response is received); or
- create a new dialog (by calling `pjsip_dlg_fork()`). In this case, subsequent responses received from this particular call leg will go to this new dialog.

Upon receiving a forked 2xx response, application can:

- decide to terminate this particular call leg. In this case, the application would construct ACK request from the response, send the ACK, then construct a BYE transaction and send it to the call-leg. Application **MUST** construct Route headers manually for both ACK and BYE requests according to the Record-Route headers found in the response before sending them to the transaction/transport layer.
- create a dialog for this particular call leg (by calling `pjsip_dlg_fork()`). Application then constructs and sends ACK request to the call leg to establish the dialog. After dialog is established, application may terminate the dialog by sending BYE request.

Application **MUST NOT** ignore a forked 2xx responses.

### Creating Forked Dialog

Application creates a forked dialog by calling `pjsip_dlg_fork()` function. This function creates a dialog and performs the following:



- Copy all attributes of the original dialog (including authorization client session) to the new dialog.
- Assign different remote tag value.
- Register the new dialog to user agent's dialog set.
- If the original dialog has an application timer, it will copy the timer and update the timer of the new dialog.

Note that the function WILL NOT copy the dialog usages (i.e. modules) from the original dialog.



The reason why the function `pjsip_dlg_fork()` doesn't copy the dialog usages from the original dialog is because each usage will normally have dialog specific data that can not be copied without knowing the semantic of the data.

After the new dialog has been created, the application then **MUST** re-register each dialog usages with the new dialog, by calling `pjsip_dlg_add_usage()`.

The new dialog then **MUST** be returned as return value of the callback function. This will cause the user agent to dispatch the message to the new dialog, causing dialog usages (e.g. application) to receive `on_rx_response()` notification on the behalf of the new dialog.

### Using Timer to Handle Failed Forked Dialog

Application can schedule application specific timer with the dialog by calling `pjsip_dlg_start_app_timer()` function. For timer associated with a dialog, this timer is preferable than general purpose timer because this timer will be automatically deleted when the dialog is destroyed.

Timer is important to handle failed forked dialog. A forked early dialog may not complete with a final response at all, because forking proxy will not forward 300-699 if it receives 2xx response. So the only way to terminate these dangling early dialogs is by setting a timer on these dialogs.

The best way to use dialog's application timer to handle failed forked early dialog, is to start the timer on the other forked dialogs the first time when it receives 2xx response on one of the dialog in the dialog set. When the timer expires and no 2xx response is received, the dialog should be terminated.

## 9.1.5 CSeq Sequencing

The local cseq of the dialog is updated when the request is sent (as opposed to when the request is created). When CSeq header is present in the request, the value may be updated as the request is sent within the dialog.

The remote cseq of the dialog is updated when a request is received. When dialog's remote cseq is empty, the first request received will set the dialog's remote cseq. For subsequent requests, when dialog receives request with cseq lower than dialog's recorded cseq, this request would be **automatically** answered statelessly by the dialog with a 500 response (Internal Server Error). When the request's cseq is greater than dialog's recorded cseq, the dialog would update the remote's cseq automatically (including when the request's cseq is greater by more than one).



This behavior is compliant with SIP specification RFC 3261 Section 12.2.2.

### 9.1.6 Authentication

The basic dialog framework provides automatic handling for authentication challenges by servers. The dialog initializes the proper authorization headers for all outgoing requests (except ACK and CANCEL), and will automatically resend the request (i.e. statefull requests) using updated authorization headers using a new transaction when it receives 401/407 response.

### 9.1.7 Stateless Operations

The dialog itself will NOT automatically create any transactions for either incoming or outgoing requests. The dialog also does NOT mandate applications to always process requests statefully, although statefull processing is strongly recommended for most requests.

Should the application decide to respond to incoming request statelessly, it must understand that future retransmissions of the same request may be answered automatically by the dialog with 500 (Internal Server Error) if another request with higher sequence number has been received by the dialog.

## 9.2 Basic UA API Reference

### 9.2.1 Dialog Creation API

A dialog can be created by calling one of the following functions.



```

pj_status_t pjsip_ua_create_uac_dlg( pjsip_user_agent *ua,
                                     const pj_str_t *local_info,
                                     const pj_str_t *local_contact,
                                     const pj_str_t *remote_info,
                                     const pj_str_t *target,
                                     pjsip_dialog **p_dlg);

```

Create a new dialog and return the instance in *p\_dlg* parameter. After creating the dialog, application can add modules as dialog usages by calling `pjsip_dlg_add_usage()`. Note that initially, the session count in the dialog will be initialized to one.



```

pj_status_t pjsip_ua_create_uas_dlg( pjsip_user_agent *ua,
                                     pjsip_rx_data *rdata,
                                     const pj_str_t *contact,
                                     pjsip_dialog **p_dlg);

```

Initialize UAS dialog from the information found in the incoming request that creates a dialog (such as INVITE, REFER, or SUBSCRIBE), and set the local Contact to *contact*. If *contact* is not specified, the local contact is initialized from the URI in the To header in the request. Note that initially, the session count in the dialog will be initialized to one.



```

pj_status_t pjsip_dlg_fork(    pjsip_dialog *original_dlg,
                              pjsip_rx_data *rdata,
                              pjsip_dialog **new_dlg );

```

Create a new (forked) dialog on receipt on forked request in *rdata*. The new dialog will be created from *original\_dlg*, except that it will have new remote tag as copied from the To header in the response. Upon return, the *new\_dlg* will have been registered to the user agent. Applications just need to add modules as dialog's usages. Note that initially, the session count in the dialog will be initialized to one.

### 9.2.2 Dialog Session Management API

The following functions are used to manage dialog's session counter.



```
pj_status_t pjsip_dlg_inc_session( pjsip_dialog *dlg );
```

Increment the number of sessions in the dialog. Note that initially (after created) the dialog already has the session counter set to one.



```
pj_status_t pjsip_dlg_dec_session( pjsip_dialog *dlg );
```

Decrement the number of sessions in the dialog. Once the session counter reach zero and there is no pending transaction, the dialog will be destroyed. Note that this function may destroy the dialog immediately if there is no pending transaction when this function is called.

### 9.2.3 Dialog Usages API

The following functions are used to manage dialog usages in a dialog.



```
pj_status_t pjsip_dlg_add_usage( pjsip_dialog *dlg,  
                                pjsip_module *module,  
                                void *mod_data );
```

Add a module as dialog usage, and optionally set the module specific data.

```
pj_status_t pjsip_dlg_set_mod_data( pjsip_dialog *dlg,  
                                    int module_id,  
                                    void *data );
```

Attach module specific data to the dialog.



```
void* pjsip_dlg_get_mod_data( pjsip_dialog *dlg,  
                              int module_id );
```

Get module specific data previously attached to the dialog.

### 9.2.4 Dialog Request and Response API



```
pj_status_t pjsip_dlg_create_request( pjsip_dialog *dlg,  
                                       const pjsip_method *method,  
                                       int cseq,  
                                       pjsip_tx_data **tdata );
```

Create a basic/generic request with the specified *method* and optionally specify the *cseq*. Use value -1 for *cseq* to have the dialog automatically put next cseq number for the request. Otherwise for some requests, e.g. CANCEL and ACK, application must put the CSeq in the original INVITE request as the parameter. This function will also put Contact header where appropriate.



```
pj_status_t pjsip_dlg_send_request ( pjsip_dialog *dlg,  
                                       pjsip_tx_data *tdata,  
                                       pjsip_transaction **p_tsx );
```

Send request message to remote peer. If the request is not an ACK request, the dialog will send the request statefully, by creating an UAC transaction and send the request with the transaction. Also when the request is not ACK or CANCEL, the dialog will increment its local cseq number and update the cseq in the request according to dialog's cseq.

If *p\_tsx* is not null, this argument will be set with the transaction instance that was used to send the request.



```
pj_status_t pjsip_dlg_send_request_stateless( pjsip_dialog *dlg,  
                                                pjsip_tx_data *tdata );
```

Send request message statelessly. The use of this function is not encouraged unless application is prepared to handle retransmissions of the message itself. However, even if the request is sent statelessly, the dialog would still update the cseq where appropriate.



```
pj_status_t pjsip_dlg_create_response( pjsip_dialog *dlg,
                                       pjsip_rx_data *rdata,
                                       int st_code,
                                       const pj_str_t *st_text,
                                       pjsip_tx_data **tdata);
```

Create a response message for the incoming request in *rdata* with status code *st\_code* and optional status text *st\_text*. This function is different than endpoint's API `pjsip_endpt_create_response()` in that the dialog function adds Contact header in the response where appropriate.



```
pj_status_t pjsip_dlg_modify_response( pjsip_dialog *dlg,
                                       pjsip_tx_data *tdata,
                                       int st_code,
                                       const pj_str_t *st_text);
```

Modify previously sent response with other status code. Contact header will be added when appropriate.



```
pj_status_t pjsip_dlg_send_response( pjsip_dialog *dlg,
                                       pjsip_tx_data *tdata,
                                       pjsip_transaction *tsx,
                                       pjsip_transaction **new_tsx );
```

Send response message *tdata* to remote statefully using a transaction.

If *tsx* argument is not NULL, the response will be sent using the transaction. If *tsx* argument is NULL, a new transaction will be created to send the response, the the transaction will be returned to caller in *new\_tsx* argument if this argument is not NULL.



```
pj_status_t pjsip_dlg_send_response_stateless( pjsip_dialog *dlg,
                                                pjsip_tx_data *tdata );
```

Send response message statelessly. Generally application doesn't want to use this function.

## 9.2.5 Dialog Auxiliary API

```
pj_status_t pjsip_dlg_set_route_set( pjsip_dialog *dlg,
                                       const pjsip_route_hdr *route_set );
```

Set dialog's (initial) route set to *route\_set* list.



```
pj_status_t pjsip_dlg_start_app_timer( pjsip_dialog *dlg,
                                       int app_id,
                                       const pj_time_val *interval,
                                       void (*cb)(pjsip_dialog*,int));
```

Start application timer with this dialog with application specific id in *app\_id* and callback to be called in *cb*. Application can only set one application timer per dialog. This timer is more usefull for dialog specific timer, because it will be automatically destroyed once the dialog is destroyed. Note that timer will also be copied to the forked dialog.



```
pj_status_t pjsip_dlg_stop_app_timer( pjsip_dialog *dlg );
```

Stop application specific timer if exists.



```
pjsip_dialog* pjsip_rdata_get_dlg( pjsip_rx_data *rdata );
```

Get the dialog instance in the incoming *rdata*. If an incoming message matches an existing dialog, the user agent must have put the matching dialog instance in the *rdata*, or otherwise this function will return NULL if the message didn't match any existing dialog.

## 9.3 Examples

### 9.3.1 Incoming Invite Dialog

The following examples uses basic/low-level dialog API to process an incoming dialog. The examples show how to:

- create and initialize incoming dialog,
- create UAS transaction to process the incoming INVITE request and transmit 1xx responses,
- transmit 2xx response to INVITE reliably,
- process the incoming ACK.

As usual, most error handlings are omitted for brevity. Real-world application should be prepare to handle error conditions in all stages of the processing.

#### Creating Initial Invite Dialog

In this example we'll learn how to create a dialog for an incoming INVITE request and respond the dialog with 180/Ringing provisional response.

```
struct app_dialog
{
    pjsip_transaction *pending_invite;
};

pj_bool_t on_rx_request(pjsip_rx_data *rdata)
{
    if (rdata->msg->line.request.method.id == PJSIP_INVITE_METHOD &&
        pjsip_rdata_get_dlg(rdata) == NULL)
    {
        // Process incoming INVITE!
        pjsip_dialog *dlg;
        pjsip_transaction *tsx;
        pjsip_tx_data *tdata;
        struct app_dialog *app_dlg;

        // Create, initialize, and register new dialog for incoming INVITE
        status = pjsip_ua_create_uas_dlg( ua, rdata, NULL, &dlg);

        // Register application as the only dialog usage
        status = pjsip_dlg_add_usage( dlg, &app_module, NULL );

        // Create 180/Ringing response
        status = pjsip_dlg_create_response( dlg, rdata, 180, NULL /*Ringing*/, &tdata);

        // Send 180 response statefully. A transaction will be created in &tsx.
        status = pjsip_dlg_send_response( dlg, tdata, NULL, &tsx);

        // As in real application, normally we will send 200/OK later,
        // when the user press the "Answer" button. In this example, we'll send
        // 200/OK in answer_dlg() function which will be explained later. In order
        // to do so, we must "save" the INVITE transaction.
        app_dlg = pj_pool_alloc( dlg->pool, sizeof(struct app_dialog));
        app_dlg->pending_invite = &tsx;
        pjsip_dlg_set_mod_data( dlg, app.mod_id, app_dlg );

        // Done processing INVITE request
        return PJ_TRUE;
    }
    // Process other requests
    ...
}
```

**Code 34 Creating Dialog for Incoming Invite**

## Answering Dialog

In this example we will learn how to send 200/OK response to establish the dialog.

```
static void answer_dlg(pjsip_dlg *dlg)
{
    struct app_dialog *app_dlg;
    pjsip_tx_data *tdata;

    app_dlg = (struct app_dlg*) pjsip_dlg_get_mod_data(dlg, app.mod_id);

    // Modify previously sent (provisional) response to 200/OK response.
    // The previously sent message is found in tsx->last_tx.
    tdata = app_dlg->invite_tsx->last_tx;
    status = pjsip_dlg_modify_response( dlg, tdata, 200, NULL /*OK*/ );

    // You may modify the response before it's sent
    // (e.g. add msg body etc).
    ...

    // Send the 200 response using previous transaction.
    // Transaction will take care of the retransmission.
    status = pjsip_dlg_send_response( dlg, tdata, app_dlg->invite_tsx, NULL );
    if (status == PJ_SUCCESS) {
        // We don't need to keep pending invite tsx anymore.
        app_dlg->invite_tsx = NULL;
    }
}
```

**Code 35 Answering Dialog**

## Processing CANCEL Request

In this example we will learn how to handle incoming CANCEL request.

```
pj_bool_t on_rx_request(pjsip_rx_data *rdata)
{
    ...
    if (rdata->msg->line.request.method.id == PJSIP_CANCEL_METHOD)
    {
        // See if we have pending INVITE transaction.
        pjsip_dialog *dlg;
        struct app_dialog *app_dlg;

        dlg = pjsip_rdata_get_dlg(rdata);
        if (!dlg) {
            // Not associated with any dialog. Respond statelessly with 481.
            status = pjsip_endpt_respond_stateless( endpt, rdata, 481, NULL, NULL,
                                                    NULL, NULL);

            return PJ_TRUE;
        }

        app_dlg = (struct app_dlg*) pjsip_dlg_get_mod_data(dlg, app.mod_id);

        if (app_dlg->pending_invite) {
            pjsip_tx_data *tdata;

            // Transaction found. Respond CANCEL (statefully) with 200 regardless
            // whether the INVITE transaction has completed or not.
            status = pjsip_endpt_respond( endpt, NULL, rdata, 200, NULL /*OK*/,
                                          NULL, NULL, NULL);

            // Create 200/OK response for CANCEL request
            status = pjsip_dlg_create_response( dlg, rdata, 200, NULL /*OK*/, &tdata);

            // Send 200/OK response to CANCEL statefully.
            status = pjsip_dlg_send_response( dlg, tdata, NULL, NULL);
        }
    }
}
```

```

        // Respond the INVITE transaction with 487/Request Terminated
        tdata = app_dlg->invite_tsx->last_tx;
        status = pjsip_dlg_modify_response( dlg, tdata, 487, NULL /*"Req Term"*/ );

        // Send the 487 response using previous transaction.
        status = pjsip_dlg_send_response( dlg, tdata, app_dlg->invite_tsx, NULL );

        if (status == PJ_SUCCESS) {
            // We don't need to keep pending invite tsx anymore.
            app_dlg->invite_tsx = NULL;
        }

    } else {
        // Transaction not found. Respond statelessly with 481
        status = pjsip_endpt_respond_stateless( endpt, rdata, 481, NULL, NULL,
                                                NULL, NULL );
    }

    // Done processing CANCEL request
    return PJ_TRUE;
}

// Process other requests
...
}

```

**Code 36 Processing CANCEL Request**

### Processing ACK Request

In this example we will learn how to handle incoming ACK request.

```

pj_bool_t on_rx_request(pjsip_rx_data *rdata)
{
    ...
    if (rdata->msg->line.request.method.id == PJSIP_ACK_METHOD &&
        pjsip_rdata_get_dlg(rdata) != NULL)
    {
        // Process the ACK request
        ...

        return PJ_TRUE;
    }
    ...
}

```

**Code 37 Processing ACK Request**

## 9.3.2 Outgoing Invite Dialog

The following sets of example demonstrate how to work with outgoing INVITE dialog.

### Creating Initial Dialog

```

static pj_status_t make_call( const pj_str_t *local_info, const pj_str_t *remote_info)
{
    pjsip_dialog *dlg;
    pjsip_method invite_method;
    pjsip_tx_data *tdata;

    // Create and initialize dialog.
    status = pjsip_ua_create_uac_dlg( user_agent, local_info, local_info,

```

```

remote_info, remote_info, &dlg );

// Register application as the only dialog usage.
status = pjsip_dlg_add_usage( dlg, &app_module, NULL);

// Send initial INVITE.
pjsip_method_set( &invite_method, &PJSIP_INVITE_METHOD );
status = pjsip_dlg_create_request( dlg, &invite_method, -1, &tdata);

// Modify the INVITE (e.g. add message body etc.. )
...

// Send the INVITE request.
status = pjsip_dlg_send_request( dlg, tdata, NULL);

// Done.
// Further responses will be received in on_rx_response.
return status;
}

```

**Code 38 Creating Outgoing Dialog**

## Receiving Response

```

static pj_bool_t on_rx_response( pjsip_rx_data *rdata )
{
    pjsip_dialog *dlg;
    dlg = pjsip_rdata_get_dlg( rdata );

    if (dlg != NULL ) {
        pjsip_transaction *tsx = pjsip_rdata_get_tsx( rdata );
        if ( tsx != NULL && tsx->method.id == PJSIP_INVITE_METHOD ) {
            if (tsx->status_code < 200)
                PJ_LOG(3,("app", "Received provisional response %d", tsx->status_code));
            else if (tsx->status_code >= 300)
                PJ_LOG(3,("app", "Dialog failed with status %d", tsx->status_code));
            else {
                PJ_LOG(3,("app", "Received OK response %d!", tsx->status_code));
                send_ack( dlg, rdata );
            }
        }
        else if (tsx == NULL && rdata->msg_info.cseq->method.id == PJSIP_INVITE_METHOD
                && rdata->msg_info.msg->line.status.code/100 == 2)
        {
            // Process 200/OK response retransmission.
            send_ack( dlg, rdata );
        }
        return PJ_TRUE;
    }
    else
        // Process other responses not belonging to any dialog
        ...
}

```

**Code 39 Receiving Response in Dialog**

## Sending ACK

```

static void send_ack( pjsip_dialog *dlg, pjsip_rx_data *rdata )
{
    pjsip_tx_data *tdata;
    pjsip_method ack_method;

    // Create ACK request
    pjsip_method_set( &ack_method, PJSIP_ACK_METHOD );
    status = pjsip_dlg_create_request( dlg, ack_method, rdata->msg_info.cseq->cseq,
                                      &tdata );
}

```



```
// Add message body
...

// Send the request.
status = pjsip_dlg_send_request_stateless( dlg, tdata, NULL );
}
```

**Code 40 Sending ACK Request**

## Chapter 10:SDP Offer/Answer Framework

The SDP offer/answer framework in PJSIP is based on RFC 3264 "An Offer/Answer Model with the Session Descriptor Protocol (SDP)". The main function of the framework is to facilitate the negotiating of media capabilities between local and remote parties, and to get agreement on which set of media to be used in one invite session.

Note that although it is mainly used by invite session, the framework itself is based on a generic SDP negotiation framework (`pjmedia_sdp_negotiator`), so it should be able to be used by other types of applications. The dialog invite session provides integration of SDP offer/answer framework with SIP protocol; it correctly interpret the message bodies in relevant messages (e.g. INVITE, ACK, PRACK, UPDATE) and translates them to SDP offer/answer negotiation.

This chapter describes the low level SDP negotiator framework, which is declared in `<pjmedia/sdp_neg.h>` header file.

### 10.1 SDP Negotiator Structure

The `pjmedia_sdp_negotiator` structure represents generic SDP offer/answer session, and is used to negotiate local's and remote's SDP.

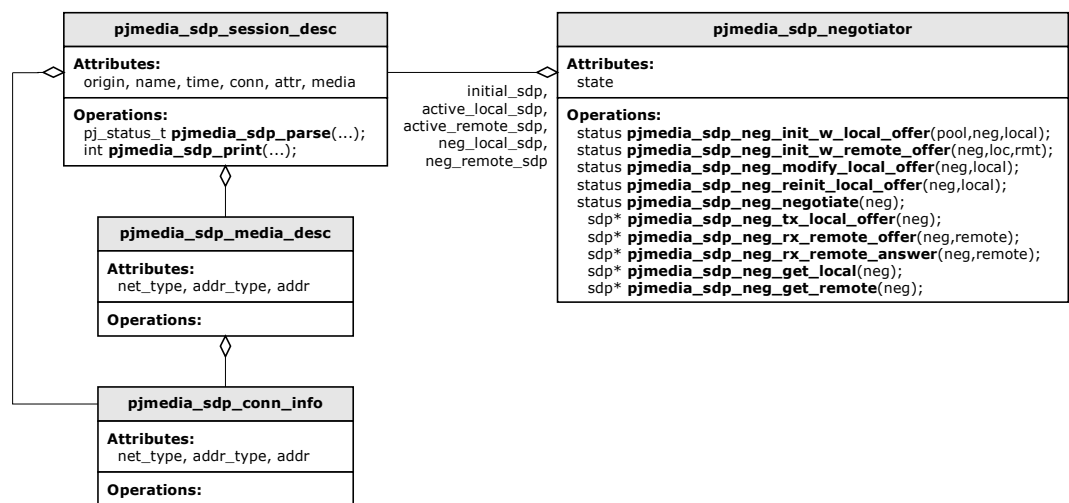


Figure 15 SDP Negotiator "Class Diagram"

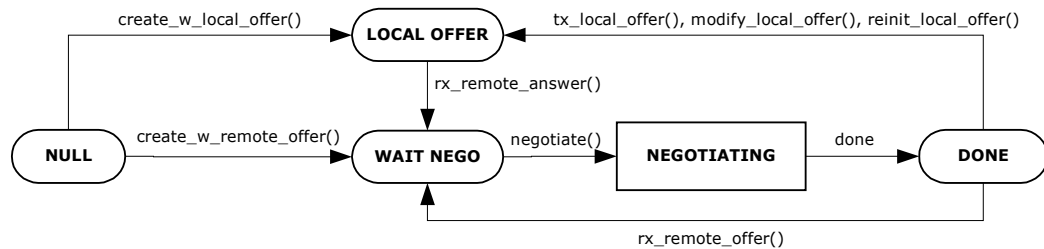
The `pjmedia_sdp_negotiator` structure keeps three SDP structures:

- **`initial_sdp`**: which is the initial capability of local endpoint. This SDP is passed to the negotiator during creation, and the contents generally will not be changed throughout the session (even after negotiation). The negotiator uses this SDP in the negotiation when it receives new offer from remote (as opposed to receiving updated SDP from remote).
- **`active_local_sdp`**: contains local SDP after it has been negotiated with remote. The dialog MUST use this to start its local media instead of the initial SDP.
- **`active_remote_sdp`**: contains the SDP currently used by peer/remote.

The negotiator also has two other SDP variables which are only used internally during negotiation process, namely `neg_local_sdp` and `neg_remote_sdp`. These are temporary SDP description, and application MUST NOT refer to these variables.

## 10.2 SDP Negotiator Session

The general state transition of SDP offer/answer session is shown in the following diagram.



**Figure 16 SDP Offer/Answer Session State Diagram**

The negotiation session starts with `PJMEDIA_SDP_NEG_STATE_NULL`. If the dialog has a local media description ready and want to offer the media to remote (normally this is the case when the dialog is acting as UAC), it creates the SDP negotiator by passing the local SDP to the function `pjmedia_sdp_neg_create_w_local_offer()`. This function will set the initial capability of local endpoint, and set the negotiation session state to `PJMEDIA_SDP_NEG_STATE_LOCAL_OFFER`. The initial SDP then can be sent to remote party in the outgoing INVITE request. Once dialog has received remote's SDP, it must call `pjmedia_sdp_neg_rx_remote_answer()` with providing the remote's SDP. The negotiation function can then be called.

If the dialog already has remote media description in hand (normally this is the case when dialog is acting as UAS), it can create the SDP negotiator session by passing both local and remote SDP to `pjmedia_sdp_neg_create_w_remote_offer()`. After this, the negotiation function can be called.

After the session has been established, both local and remote party may modify the session. The negotiator can handle one of these two situations:

- The dialog has received SDP from remote. In this case, the dialog will call `pjmedia_sdp_neg_rx_remote_offer()` and passing the remote's SDP to this function. After this the negotiation function can be called. The negotiation function's return value determines whether there is modification needed in the local media.
- The local party wants to send SDP to remote. Dialog can further choose one of the following actions:
  - If it just wants to send currently active local SDP without modification, it should call `pjmedia_sdp_neg_tx_local_offer()` to get the active local SDP, send the SDP, then wait for the remote's answer.
  - If it wants to modify currently active local media (e.g. changing stream direction, change active codec, etc), it should get the active local media with `pjmedia_sdp_neg_get_local()`, modify it, call `pjmedia_sdp_neg_modify_local_offer()` to update the offer, send the local SDP, then wait for the remote's answer.

- The dialog may want to completely change the local media (e.g. changing IP address, changing codec set, adding new media line). This is different than updating current media described above because it will change `initial_sdp`, so that future negotiation will be based on this new SDP. If the dialog wants to do this, it calls `pjmedia_sdp_neg_reinit_local_offer()` with the new local SDP, send the SDP, then wait for remote's answer.

After the dialog has sent offer to remote party, it should receive answer back from the remote party. The dialog must provide the remote's SDP to the negotiator so that the negotiation function can be called. The dialog provides the remote's answer by calling `pjsip_sdp_neg_rx_remote_answer()`.

If remote has rejected local's offer (e.g. returning 488/"Not Acceptable Here" response), dialog MUST still call `pjsip_sdp_neg_rx_remote_answer()` with providing NULL in remote's SDP argument, and call the negotiation function so that the negotiator session can revert back to previously active session descriptions, if any.

## 10.3 SDP Negotiation Function

The dialog calls `pjmedia_sdp_neg_negotiate()` to negotiate the offer and the answer, after it has provided both local's and remote's SDP to be used for the negotiation (i.e. negotiator state is `PJMEDIA_SDP_NEG_STATE_WAIT_NEGO`). This function may return one of the following result:

- `PJ_SUCCESS`, (i.e. zero) if it has successfully established an agreement between local and remote SDP. In this case, both local's and remote's *active* SDP will be stored in the session for future reference, and application can query these active SDPs to start the local media.
- `PJMEDIA_ESDPNOCHANGE`, if it found out that there is no modification needed in currently used SDPs (both local and remote). In this case, the previously agreed SDP sessions will not be modified either.
- `PJMEDIA_ESDPFAIL`, if it couldn't find agreement on local and remote capabilities. In this case, if the session is keeping a previously agreed SDP, these SDP (local and remote) will not be modified. If dialog is acting as UAS for this session, it should respond the request with 488/Not Acceptable Here response to the offer.
- `PJMEDIA_ESDPNOOFFER`, if negotiator has not sent/received any offer yet.
- `PJMEDIA_ESDPNOANSWER`, if negotiator has not received remote's answer yet.
- or other non-zero value to indicate other errors.

In all cases, the negotiation function will set the negotiator's state to `PJMEDIA_SDP_NEG_STATE_DONE`.

## Chapter 11: Dialog Invite Usage

**(DRAFT, TO BE DONE)**

### 11.1 Introduction

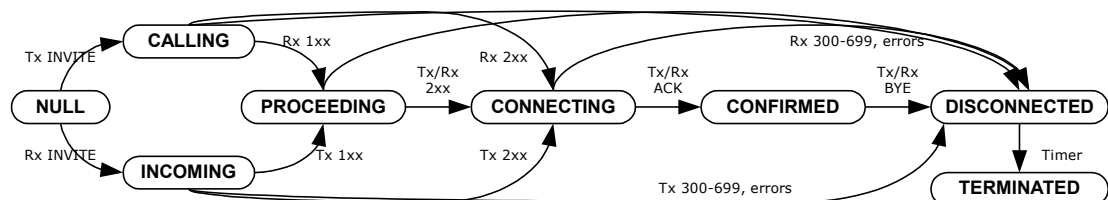
The dialog invite usage is a module which can be registered to dialog to provide higher level invite session processing. This module provides the following functionalities to application:

- Session progress reporting (e.g. call progressing, connected, confirmed, disconnected)
- SDP offer and answer framework,
- Session/call transfer,
- High-level forking handler,
- Session offer timeout (i.e. Expires header),
- Session extensions, such as session timer, and reliable provisional response.

#### 11.1.1 Invite Session State

The dialog invite usage provides callback to notify application about session progress. This is particularly useful for telephony applications, where the session's state is normally associated with telephony call state.

The progress of an invite session is defined with the following state:



**Figure 17 Invite Session State Diagram**

The description of each state is as follows:

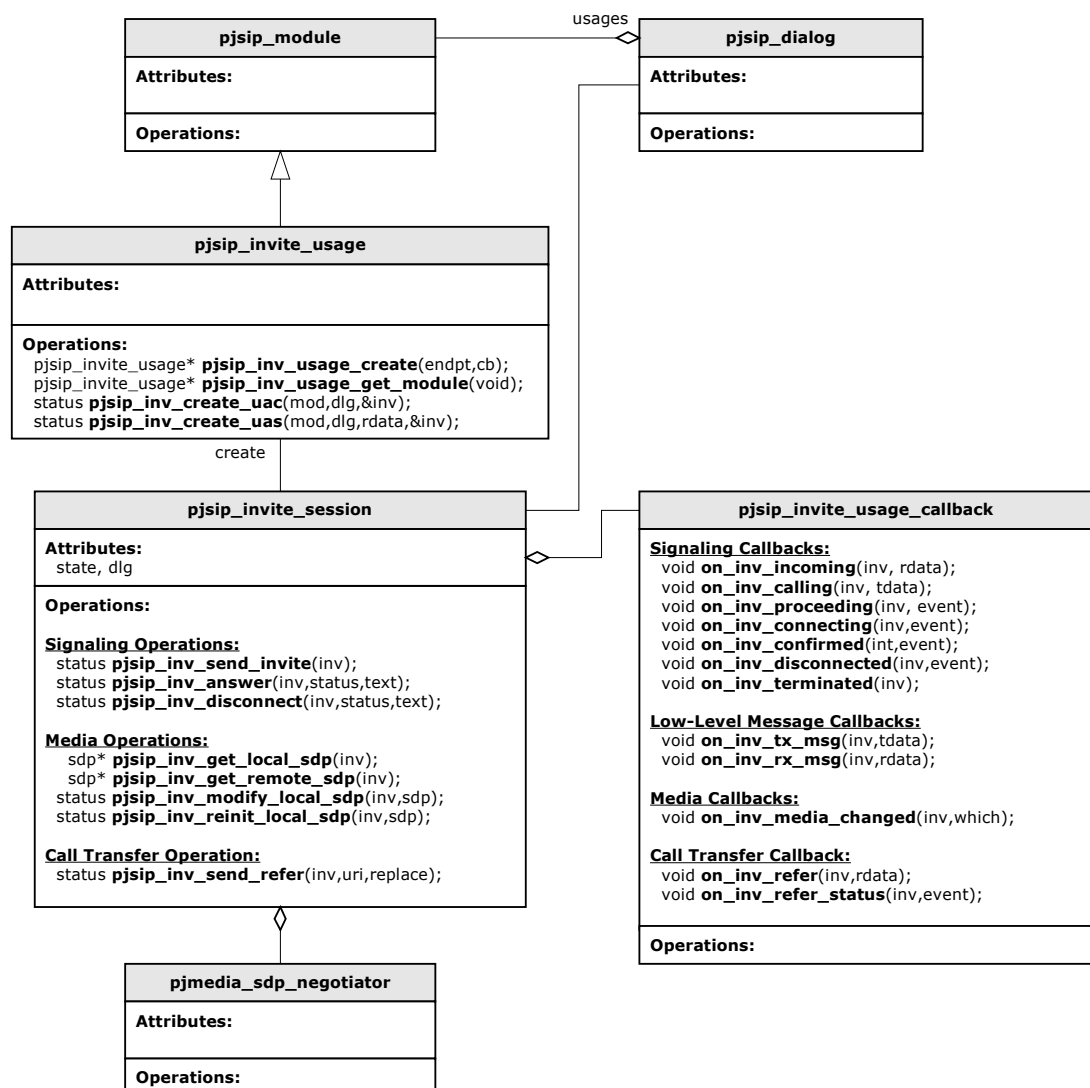
PJSIP_INV_STATE_NULL	This is the state of the session when it was first created. No messages have been sent/received at this point.
PJSIP_INV_STATE_CALLING	The session state after the first INVITE message is sent, but before any provisional response is received.
PJSIP_INV_STATE_INCOMING	The session state after the first INVITE message is received, but before any provisional response is sent.
PJSIP_INV_STATE_PROCEEDING	The session state after dialog has sent or received provisional response messages for the INVITE request, including 100/Trying response.
PJSIP_INV_STATE_CONNECTING	The session state after a final 2xx response has

	been sent or received.
<b>PJSIP_INV_STATE_CONFIRMED</b>	The session state after ACK request has been sent or received.
<b>PJSIP_INV_STATE_DISCONNECTED</b>	The session state when the session has been disconnected, either because of non-successful final response to INVITE or BYE request.
<b>PJSIP_INV_STATE_TERMINATED</b>	The session state when the session will be destroyed. All resources will be freed when this state is reached.

**Figure 18 Invite Session State Description**

### 11.1.2 Invite Usage "Class Diagram"

The following figure shows the invite usage class diagram.

**Figure 19 Dialog Invite Usage "Class Diagram"**

## Chapter 12:Dialog Subscribe Usage

---

**(TO BE DONE)**