Technical Manual

1. Introduction

pjsua is an open source command line SIP user agent (softphone) that is used as the reference implementation for PJSIP, PJNATH, and PJMEDIA. Despite its simple command line appearance, it does pack many features!

SIP features:

- Mutiple lines/identities (account registrations).
- Multiple calls.
- PRACK (100rel, RFC 3262).
- UPDATE (RFC 3311).
- OPTIONS.
- Call hold.

Media features:

- Multiple Concurrent calls
- Conferencing

NAT traversal features:

1. ICE (Interactive Connectivity Establishment, latest ICE draft).
2. STUN (latest rfc3489-bis).

2. Reference

Synopsis

The options value can be specified after the option with either whitespace or equal sign.

```
Usage:
  pjsua [options] [SIP URL to call]

General options:
  --config-file=file  Read the config/arguments from file.
  --help              Display this help screen
  --version           Display version info

Logging options:
  --log-file=fname    Log to filename (default stderr)
  --log-level=N       Set log max level to N (0(none) to 6(trace)) (default=5)
```

```
  --app-log-level=N   Set log max level for stdout display (default=4)
```

PJSUA can be configured with zero or more SIP accounts. SIP accounts can be used to log in to SIP services, and send or receive requests using the specified SIP services.

SIP Account options:

```
  --use-ims           Enable 3GPP/IMS related settings on this account
  --use-srtp=N        Use SRTP?  0:disabled, 1:optional, 2:mandatory (def:0)
  --srtp-secure=N     SRTP require secure SIP? 0:no, 1:tls, 1:sips (def:1)
  --registrar=url     Set the URL of registrar server
  --id=url            Set the URL of local ID (used in From header)
  --contact=url       Optionally override the Contact information
  --proxy=url         Optional URL of proxy server to visit
                      May be specified multiple times
  --reg-timeout=SEC   Optional registration interval (default 55)
  --realm=string      Set realm
  --username=string   Set authentication username
  --password=string   Set authentication password
  --publish           Send presence PUBLISH for this account
  --use-100rel        Require reliable provisional response (100rel)
  --auto-update-nat=N Where N is 0 or 1 to enable/disable SIP traversal behind
                      symmetric NAT (default 1)
  --next-cred         Add another credentials
```

Each account has separate authentication settings, and you can put multiple credentials in one account (e.g. when one need to specify different credentials for each proxies)

SIP Account Control:

```
  --next-account      Add more account
```

Transport Options:

```
  --local-port=port   Set TCP/UDP port. This implicitly enables both
                      TCP and UDP transports on the specified port, unless
                      if TCP or UDP is disabled.
  --ip-addr=IP        Use the specifed address as SIP and RTP addresses.
                      (Hint: the IP may be the public IP of the NAT/router)
  --no-tcp            Disable TCP transport.
  --no-udp            Disable UDP transport.
  --nameserver=NS     Add the specified nameserver to enable SRV resolution
                      This option can be specified multiple times.
  --outbound=url      Set the URL of global outbound proxy server
                      May be specified multiple times
  --stun-srv=name     Set STUN server host or domain
```

Media Options:

```
  --add-codec=name    Manually add codec (default is to enable all)
  --dis-codec=name    Disable codec (can be specified multiple times)
  --clock-rate=N      Override conference bridge clock rate
```

```
  --snd-clock-rate=N  Override sound device clock rate
  --stereo            Audio device and conference bridge opened in stereo mode
  --null-audio        Use NULL audio device
  --play-file=file    Register WAV file in conference bridge.
                      This can be specified multiple times.
  --play-tone=FORMAT  Register tone to the conference bridge.
                      FORMAT is 'F1,F2,ON,OFF', where F1,F2 are
                      frequencies, and ON,OFF=on/off duration in msec.
                      This can be specified multiple times.
  --auto-play         Automatically play the file (to incoming calls only)
  --auto-loop         Automatically loop incoming RTP to outgoing RTP
  --auto-conf         Automatically put calls in conference with others
  --rec-file=file     Open file recorder (extension can be .wav or .mp3
  --auto-rec          Automatically record conversation
  --quality=N         Specify media quality (0-10, default=6)
  --ptime=MSEC        Override codec ptime to MSEC (default=specific)
  --no-vad            Disable VAD/silence detector (default=vad enabled)
  --ec-tail=MSEC      Set echo canceller tail length (default=256)
  --ilbc-mode=MODE    Set iLBC codec mode (20 or 30, default is 20)
  --capture-dev=id    Audio capture device ID (default=-1)
  --playback-dev=id   Audio playback device ID (default=-1)
  --capture-lat=N     Audio capture latency, in ms (default=100)
  --playback-lat=N    Audio playback latency, in ms (default=100)
  --snd-auto-close=N  Auto close audio device when it is idle for N seconds.
                      Specify N=-1 (default) to disable this feature.
                      Specify N=0 for instant close when unused.
  --no-tones          Disable audible tones
```

Media Transport Options:

```
  --use-ice           Enable ICE (default:no)
  --ice-no-host       Disable ICE host candidates
  --rtp-port=N        Base port to try for RTP (default=4000)
  --rx-drop-pct=PCT   Drop PCT percent of RX RTP (for pkt lost sim, default: 0)
  --tx-drop-pct=PCT   Drop PCT percent of TX RTP (for pkt lost sim, default: 0)
  --use-turn          Enable TURN relay with ICE (default:no)
  --turn-srv          Domain or host name of TURN server ("NAME:PORT" format)
  --turn-tcp          Use TCP connection to TURN server (default no)
  --turn-user         TURN username
  --turn-passwd       TURN password
```

Buddy List (can be more than one):

```
  --add-buddy url     Add the specified URL to the buddy list.
```

User Agent options:

```
  --auto-answer=code  Automatically answer incoming calls with code (e.g. 200)
  --max-calls=N       Maximum number of concurrent calls (default:4, max:255)
  --thread-cnt=N      Number of worker threads (default:1)
  --duration=SEC      Set maximum call duration (default:no limit)
  --norefersub        Suppress event subscription when transfering calls
  --use-compact-form  Minimize SIP message size
```

```
When URL is specified, pjsua will immediately initiate call to that URL
```

3. Application Menus

These command line menus are available within the application. To invoke the command, input the command then press ENTER.

```
+=============================================================================+
|       Call Commands:         |   Buddy, IM & Presence:  |     Account:      |
|                              |                          |                   |
|  m  Make new call            | +b  Add new buddy       .| +a  Add new accnt |
|  M  Make multiple calls      | -b  Delete buddy         | -a  Delete accnt. |
|  a  Answer call              |  i  Send IM              | !a  Modify accnt. |
|  h  Hangup call  (ha=all)    |  s  Subscribe presence   | rr  (Re-)register |
|  H  Hold call                |  u  Unsubscribe presence | ru  Unregister    |
|  v  re-inVite (release hold) |  t  ToGgle Online status |  >  Cycle next ac.|
|  U  send UPDATE              |  T  Set online status    |  <  Cycle prev ac.|
| ],[ Select next/prev call    +--------------------------+-------------------+
|  x  Xfer call                |      Media Commands:     |  Status & Config: |
|  X  Xfer with Replaces       |                          |                   |
|  #  Send RFC 2833 DTMF       | cl  List ports           |  d  Dump status   |
|  *  Send DTMF with INFO      | cc  Connect port         | dd  Dump detailed |
| dq  Dump curr. call quality  | cd  Disconnect port      | dc  Dump config   |
|                              |  V  Adjust audio Volume  |  f  Save config   |
|  S  Send arbitrary REQUEST   | Cp  Codec priorities     |  f  Save config   |
+------------------------------+--------------------------+-------------------+
|  q  QUIT       sleep MS     echo [0|1|txt]        n: detect NAT type      |
+=============================================================================+
```

Call Commands

m	Make new call	Make a new call/INVITE. The application will ask the URL of the remote peer to contact.
M	Make multiple calls	Make multiple calls to the same destination.
a	Answer call	Send 100-699 response to current call. The application will ask which status code to send. Note that current call MUST be an incoming call. Current call can be selected with "]" or "[" command.
h	Hangup call	Hangup current call. This command will work regardless of the state of the current call (e.g. it may send CANCEL, 603 (Decline), BYE, etc depending on the state of the call).
H	Hold call	Put the current call on-hold by sending inactive SDP. Note that incoming call hold request will be acted automatically.
v	Re-Invite (release	Send active SDP with current call. If the call is currently on-hold, this will

	hold)	effectively release the hold. You can also change the local codec preference with **Cp** command before sending the offer.
U	Send UPDATE request	Send UPDATE with new offer. You can also change the local codec preference with **Cp** command before sending the offer.
]	Select next call	If application has more than one calls, this command will select the next call in the list as current call.
[	Select previous call	If application has more than one calls, this command will select the previous call in the list as current call.
x	Transfer call (xfer)	Transfer current call (i.e. send outgoing REFER). The application will ask the URL to which remote party should contact. Note that transfering current call DOES NOT cause pjsua to hold or disconnect currentcall. User should use the hold and hangup command to hold and terminate the call accordingly. Note that incoming call transfer request will be processed automatically.
#	Send DTMF with RFC 2833	Send DTMF digits as RFC 2833 events in current call. The application will ask the digit strings to send.
*****	Send DTMF with SIP INFO	Send DTMF digits as SIP INFO for current call. The application will ask the digit strings to send.
dq	Dump (call) quality	Print media statistic (packet loss, duplicate, jitter, end-to-end delay, etc) of currently selected call.
S	Send arbitrary request	Send an arbitrary request to remote host. You will be asked about the SIP method and destination to send the request. Useful for example to send OPTIONS.

IM and Presence Commands

+b	Add buddy	Add a new buddy URL to the buddy list.
-b	Delete buddy	Delete a buddy from the buddy list.
i	Send IM	Send outgoing MESSAGE. The application will ask the URL of the remote peer to send the message to, and the contents of the message.
s	Subscribe presence	Subscribe to presence subscription of an URL in the buddy list. The buddy's online presence status will be monitored by the application.
u	Unsubscribe presence	Unsubscribe existing presence subscription.
t	Toggle online state	Toggle local presence's online status. If there are subscribers to our presence, NOTIFY messages will be sent to those subscribers. Note that application automatically accepts presence subscription request.
T	Specify custom presence text	Specify enhanced presence status text (such as "Be Right Back") with this command.

Account Commands

+a	Add account	Add a new account (not implemented yet).
-a	Delete account	Delete account (not implemented yet).
!a	Modify account	Modify account (not implemented yet).
rr	Re-Register	Send REGISTER request for this account to register or to refresh registration.
ru	Unregister	Send REGISTER request to unregister the account registration.
>	Select next account	Select the current account to be used for sending outgoing requests.
<	Select prev account	Select the current account to be used for sending outgoing requests.

Conference Commands

cl	Conference List	List all the ports registered to the conference bridge, and show the interconnection among these ports.
cc	Conference Connect	Create a unidirectional connection between two ports. For example, if you have a WAV player connected at slot #1 and a call connected at slot #2, you can stream WAV file to the call by specifying this command: **`cc 1 2`**.
cd	Conference Disconnect	Disconnect a unidirectional connection between two ports. Example: `cd 1 2`.
V	Adjust volume	Make adjustment to the audio level of a particular media port.
Cp	Arrange codec priorities	Arrange the codec priorities. Useful for example to set the prefered codec before sending re-INVITE ("v" command) or UPDATE ("U" command).

Status and Config Commands

d	Dump status	Dump the contents of endpoint, transaction table, dialog table, invite sessions, etc to the screen.
dd	Dump detailed status	Dump detailed status (each transaction, each call, including call/media quality etc.)
dc	Dump configuration	Dump current configuration to screen.
f	Write settings	Write current configuration to file.

Other Commands

<table>
<tr><td>q</td><td>Quit</td><td>Quit application. All current calls, subscriptions, and registrations will be terminated.</td></tr>
<tr><td>sleep MSEC</td><td>Suspend keyboard input</td><td>Suspend keyboard input for the specified miliseconds. Useful when piping commands to pjsua.</td></tr>
<tr><td>echo [0|1|TXT]</td><td>Control command echo</td><td>Use <code>echo 0</code> or <code>echo 1</code> to disable or enable command echo (default is disabled). Use <code>echo TXT</code> (where TXT is any text) to output the text to stdout.</td></tr>
</table>

n	Detect network type	Initiate NAT type detection. The result will be printed to stdout and log.

4. References

1. www.symbian.org.
2. www.developer.uiq.com
3. www.newlc.com
4. www.allaboutsymbian.com
5. www.s60.com