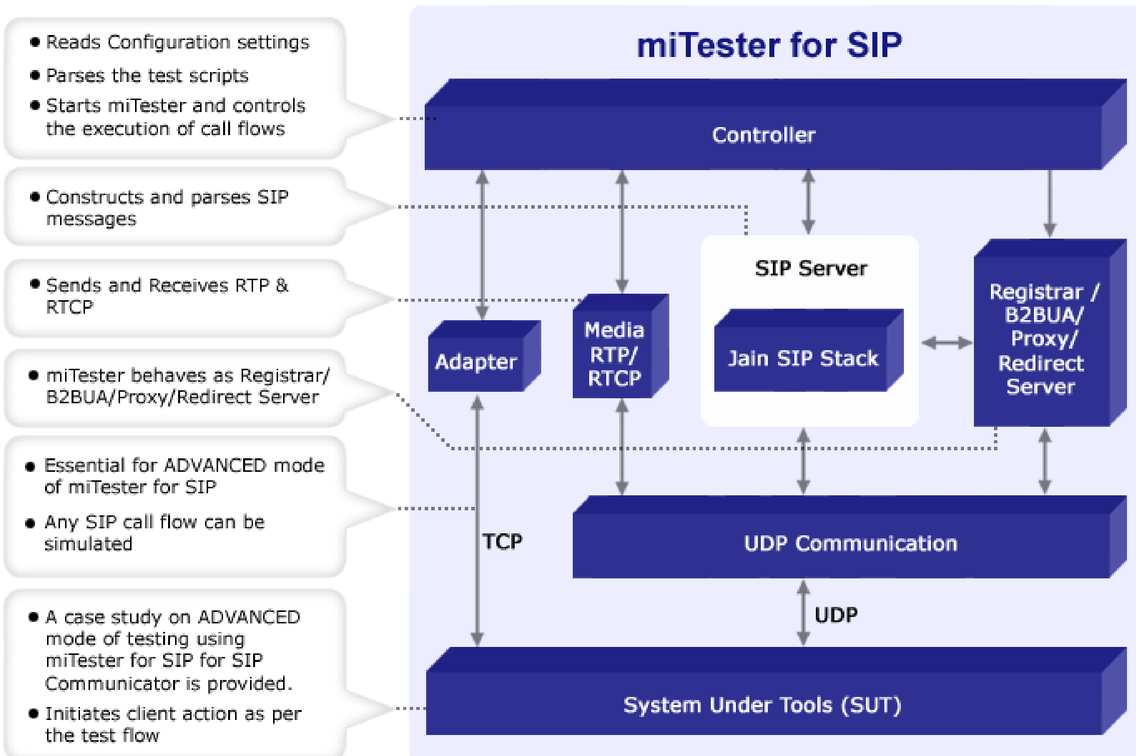
mitesterforsip_Architecture

Release Version 2.0

mitester for SIP - Architecture

mitester for SIP automates any SIP application call flows and simulates as a sip server to test. It uses JAIN SIP stack for parsing and framing of SIP messages. It works either in **USER** or **ADVANCED** mode. In the ADVANCED mode, it executes both client and server test scripts there are no manual intervention in the test execution.



Controller

Controller reads the configuration settings. It uses JAXB Parser to parse and validate the test scripts (client and server XML scripts) according to the XML schema defined.

Controller module controls the execution of test scripts. It controls both client and server test execution in **ADVANCED** mode and controls the server test execution in **USER** mode.

Control actions are defined in the test scripts. It implements timer threads which are used to suspend and maintain the flow of test executions.

SIP Server

SIP Server is the main module of mitester. It uses JAIN SIP stack, SIP Framer, SIP Parser and UDP listener components. This module uses JAIN SIP stack for constructing and parsing SIP messages.

SIP Server supports the following activities at the server test execution

- Parsing received SIP messages
- Framing the SIP Request or SIP Response according to the server scenario with mandatory headers (Request line or Status line, From, To, CallID, CSeq, Contact, MaxForwards, Via and ContentLength) only.
- Adding other than mandatory headers (Optional Headers) defined in the RFC 3261
- Supports Call Forking (Generating request or response messages with user-specified dialog, irrespective of the current dialog).
- Supports Load Testing (Test scenarios can be executed 'n' number of times).
- Also supports Abnormal test case execution (Fault-Injection testing)
    - missing mandatory headers
    - Duplication of SIP headers, header parameters and content
    - Duplication of SIP messages
    - Adding CRCR / LFLF / CRLF to any particular header or all headers present in the SIP messages
    - Missing CR / LF / CRLF to any particular header or all the headers present in the SIP message

Media

Module supports RTP and RTCP call flow simulation. It sends and receives RTP and RTCP packets through the UDP communication interface. It uses the jnetstream for fetc hing media packets from pcap file.

Adapter

Adapter module establishes TCP channel between the System Under Test (SUT) and mitester. Through this TCP Channel. Adapter communicates the SUT and initiates client action as per the test script.

System Under Test (SUT)

System Under Test is the SIP Application that we test.

As a case study, sip communicator is chosen as system under test (SUT). An interface is developed for automating the test executions of the SUT.

Registrar/B2BUA/Proxy/ /Redirect Server

Module defines the basic SIP Server features.

- Registrar
- B2BUA
- PROXY
- Redirect Server

Registrar
mitester receives the REGISTER SIP Message from UDP channel and sends 200 or 401 to the Register request.

B2BUA:
mitester receives SIP messages from UDP channel.It act as both client & Server and processes the Messages.

PROXY:
mitester receives SIP messages from UDP channel.It adds Via & Record-Route headers to the SIP Request Messages and removes top most via headers and forwards the SIP Response Messages to UDP channel.

Redirect server:
mitester receives INVITE Response with Status Code 302 from UDP Channel. It reconstructs the New INVITE Request and sends to the UDP channel.