

# SIPStudio

Addition to SIPSim  
SIP-based services  
simulation

## SIPStudio: Enhancing SIPSim

SIPStudio is a sophisticated addition to RADCOM's SIPSim that provides extended flexibility in both call flow and scenario generation for the SIPSim core application.

The SIPStudio retains the highest call load figures of the SIPSim while still handling an infinite number of different calls flows and programmable behaviors.

As the SIP protocol becomes more widely used, the number of applications using SIP continues to grow. These applications use SIP flows that are much more complicated than regular VoIP call flows. SIPStudio is specially designed to allow description tailoring and execution of these SIP flows under huge loads, as dictated by the regular SIPSim. SIPStudio greatly simplifies the creation, generation and monitoring of complicated SIP flows and combines several flows into a complex scenario of a single run.

## Combining Performance and Flexibility in a Single Machine

Fundamental to SIPStudio is its ability to combine full scriptable operation with the largest performance numbers, in terms of both the number of concurrent calls that can be held open and the number of calls per second opened by the generator.

The advanced design of SIPStudio allows it to program any desired call flow into the operation while generating up to 3,000 flows per second and keeping up to 500,000 of these flows open at any given moment. This level of performance is unprecedented, with rival SIP call generators normally required to make huge compromises on performance in order to run slightly different call flows on their machines.

The benefits of such flexibility and proven performance are evident when testing medium and complex scenarios: a single SIPStudio can replace many competing units for all types of testing, meaning huge savings on capital expenses.

## SIPStudio Highlights

- ▶ **Call flow control:** With the new SIPSim XML-based call flow engine, control individual call flows allowing for every imaginable SIP call behavior
- ▶ **Simple GUI:** New, simpler SIPStudio GUI
- ▶ **Scenario options:** Newly formed call flow and scenario capabilities provide a simple hierarchical approach to managing and activating the high load into the UUT
- ▶ **Same high performance:** With added flexibility and new features, still has the same high performance of the older SIPSim
- ▶ **Multiple codecs:** The new SIPStudio infrastructure easily delivers multiple codecs
- ▶ **Simple construction of complex scenarios:** Very simple, Excel-like composition of simulation scenarios
- ▶ **Improved session statistics:** Much finer resolution for session statistics.
- ▶ **Easy setup:** Configuration of SIP phones and SIP destinations has been simplified

## Target Users

**Network Element Developers:** SBC; signaling; gateway; softswitches

**Design and Verification Labs:** QA, performance or development labs

## User Environments

VoIP technologies, IMS technologies, 3G technologies

Voice, Video, Instant Messaging (IM), Presence

## Supported RFCs, Methods and Protocols

INVITE, BYE, ACK, CANCEL, REGISTER, OPTIONS, NOTIFY, SUBSCRIBE, REFER INFORMATION, UPDATE, MESSAGE, PUBLISH, PRACK

## Specifications

### R70, Rack-mount 2U

Number of LIMs: up to 3  
 Width: 430mm  
 Height: 89mm (2U)  
 Depth: 540mm  
 Weight: 9.5 Kg

### R1000, Rack-mount 2U

Number of LIMs: up to 2  
 Width: 440mm  
 Height: 89mm (2U)  
 Depth: 475mm  
 Weight: 9.0 Kg

## Performance

### Signaling

Call open rate: 3,000 calls/sec  
 Simultaneous calls: 500,000  
 BHCA: 6,000,000

### Signaling + media

Call open rate: 1,000  
 Simultaneous calls: 50,000 (500,000 with signaling)  
 BHCA: 3,000,000

The screenshot shows the SIPStudio interface. On the left is a tree view with categories: SIP Phones, Flows (containing IMS flow1, regular call, DDOS scenario A1, voice\_IM, Presence), Models (containing slow ramp, fast ramp, Presence), Destinations Lists (containing IMS users, 3G users), Media Profiles, and Scenarios (containing audio\_presense\_IM). The main window is titled 'Flow - regular call' and contains the following fields:

- Flow File Name: Register
- Flow Description: The registration script defines a registration procedure between a SIP user and the registrar server. You can configure the script to refresh registration if a 200 OK response is received, the script uses the expiration value included in the 200 OK response. The script handles the following:
- Flow parameters table:

Parameter Name	Value	Description
Expires_Header	3600	Defines an Expires Header value to include with the
Contact_Header_Expire_P...	3600	Defines the expiration for the Contact header. If t
Do_Refresh_Registration	true	If configured to True, the script refreshes the Reg

Buttons include 'Customize Flow Messages', 'Reset To Default', and 'Apply'. The bottom status bar shows 'Simulation Setup', 'General Statistics', 'Signaling Statistics', 'Flow Statistics', 'History Statistics', 'RTP Statistics', and a timer at '00:00:00'.

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