

# OpenSIPS Virtual Machine

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## 1. Prerequisites :

- ★ OpenSIPS - VM archive downloaded (tgz format) and unpacked
- ★ VM Ware player – to run the virtual machine.
- ★ import your OpenSIPS VM into the VMware player

## 2. How to access VM

- ★ you need to place the OpenSIPS VM into a network where a DHCP server is available; by default at startup, OpenSIPS VM will try to get an IP via DHCP
- ★ you can connect to the OpenSIPS VM using SSH . Access info is: user: root, password: opensips

## 3. System Config via OpenSIPS CP

### 3.1 Access OpenSIPS CP

- ★ to access OpenSIPS CP web interface use your browser and type `http://vm_ip_address/cp`
- ★ for login into Control Panel (CP) interface use `admin/admin`

### **3.2 Configure Domains**

- ★ Use System → Domains tool: to manage the SIP domains settings
- ★ if you want to change your sip domain feel free to do it (default it's **test.com** – for testing purposes)

### **3.3 Configure Dialplan**

- ★ Use System → Dialplan tool : to manage the dialplan settings
- ★ as you can see there are two predefined rules there – one for local users and one for pstn users (see the attributes)
- ★ if you want to change the “meaning” of the two rules (for example, you want to edit the regex that identifies a “local” user, or you want your pstn numbers to stat with something different than 00) you can do it but remember to keep the Attributes names the same (pstn & local)
- ★ you can add more than one rule for each of these two attributes, but remember that the dialplan id must be 0

### **3.4 Configure Dynamic Routing**

- ★ Use System → Dynamic Routing tool : to change the way your calls are routed
- ★ if you want to send your calls to PSTN you have to add some gateways
- ★ when adding a new gateway be careful that the Pri Prefix matches at least one on the rules for pstn in Dialplan

## **4. Users Management via OpenSIPS CP**

### **4.1 Adding users**

- ★ Use Users → Users Management to add or remove users
- ★ to add an user click Add New and fill in the fields
- ★ do not forget that the names of the users must match at least one of the rules for “local” in Dialplan

### **4.2 ACL Management**

- ★ Use Users → ACL Management to asign each user a group (for example, user “user” is in group “pstn” - so he is allowed to call PSTN)

★ a user can be in more than one group (for example, you can add a new ACL and put user “user” in group “voicemail” - this feature is not available in this version – and he will have voicemail features)