



## Configuration Guide for Voiceflex SIP Trunks on Cisco UC 540 Phone System

version 0.1  
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**Please read the following information carefully before making any changes on your Live-PBX**

All information in this document is for example purposes only and was created from the devices in a specific lab environment. **All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.** For actual parameters for your SIP trunk, please refer to your SIP Configuration sheet (provided with all VoiceFlex SIP trunks) for all usernames, passwords, IP addresses and settings relevant to each trunk. This can be downloaded from the VoiceFlex web-portal or requested from the support team. Configuration of VoiceFlex SIP trunks on these devices should be undertaken only by technicians trained in, and familiar with, the Cisco System.

**Introduction:**

The information in this document is intended as a general guide to help you set up the Voiceflex service on your SIP compatible telephone system. It contains only Voiceflex-specific parameters that need to be configured on your IP-PBX. For full and proper functioning of the Voiceflex service and all other features, your IP-PBX should be set up according to the detailed instructions provided by the manufacturer in their configuration guide

**Network Topology:**

*For your network settings, we assume that you already define port forwarding for UDP 5060 on your Firewall/Router. If your system doesn't allow activating port forwarding then please consult Voiceflex support team for possible options which can solve your problem.*

***All security and NAT configurations are configured on 877 DSL router, so there is not any NAT or Firewall configuration on Cisco UC-540, so all the security and NAT parameter are disabled because of the network topology. Please be aware that Voiceflex doesn't encourage their customer to disable, deactivate their security configuration. It is customer own risk to disable any configured security parameter.***

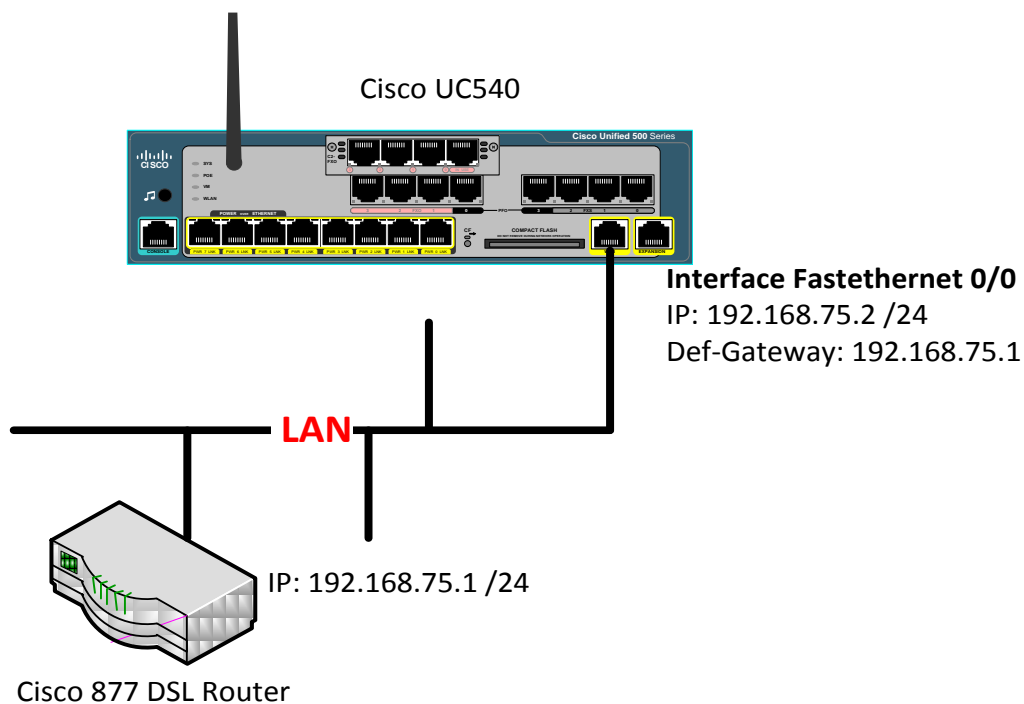


Figure-1: Network Diagram

**Setup a new SIP Trunk on UC540:**

Sample SIP Configuration Parameters

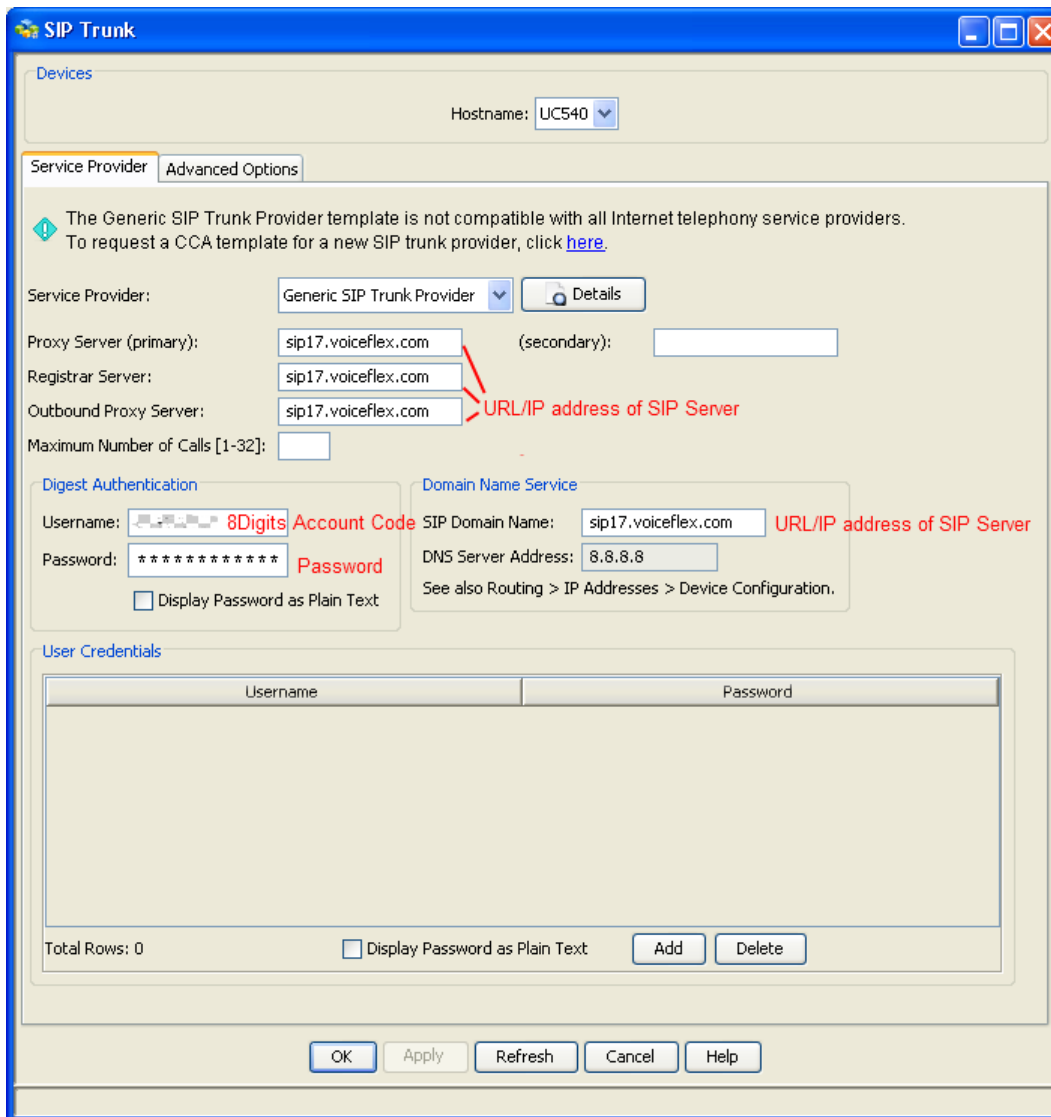
SIP User: **8Digits Account Code**

Password: XXXXX

SIP Proxy / Registrar: sip17.voiceflex.com / 93.95.124.76

1. Launch CCA and connect to the UC500.
2. Choose Configuration --> Telephony --> Voice --> SIP trunk tab (Note with CCA 2.1 or later, the SIP trunk tab will move to Configuration > Telephony > Trunks > SIP Trunks)

Choose Generic SIP Trunk Provider and fill the gaps based on the screen below.



Proxy Server (Primary) : **sip17.voiceflex.com / sip11.voiceflex.com / sip.voiceflex.com**

Registrar Server : **sip17.voiceflex.com / sip11.voiceflex.com / sip.voiceflex.com**

Outbound Proxy Server : **sip17.voiceflex.com / sip11.voiceflex.com / sip.voiceflex.com**

Depends on your SIP trunks settings, you can choose one of the server



Digest Authentication:

Username : **8Digits Account Code**

Password : **SIP-Password**

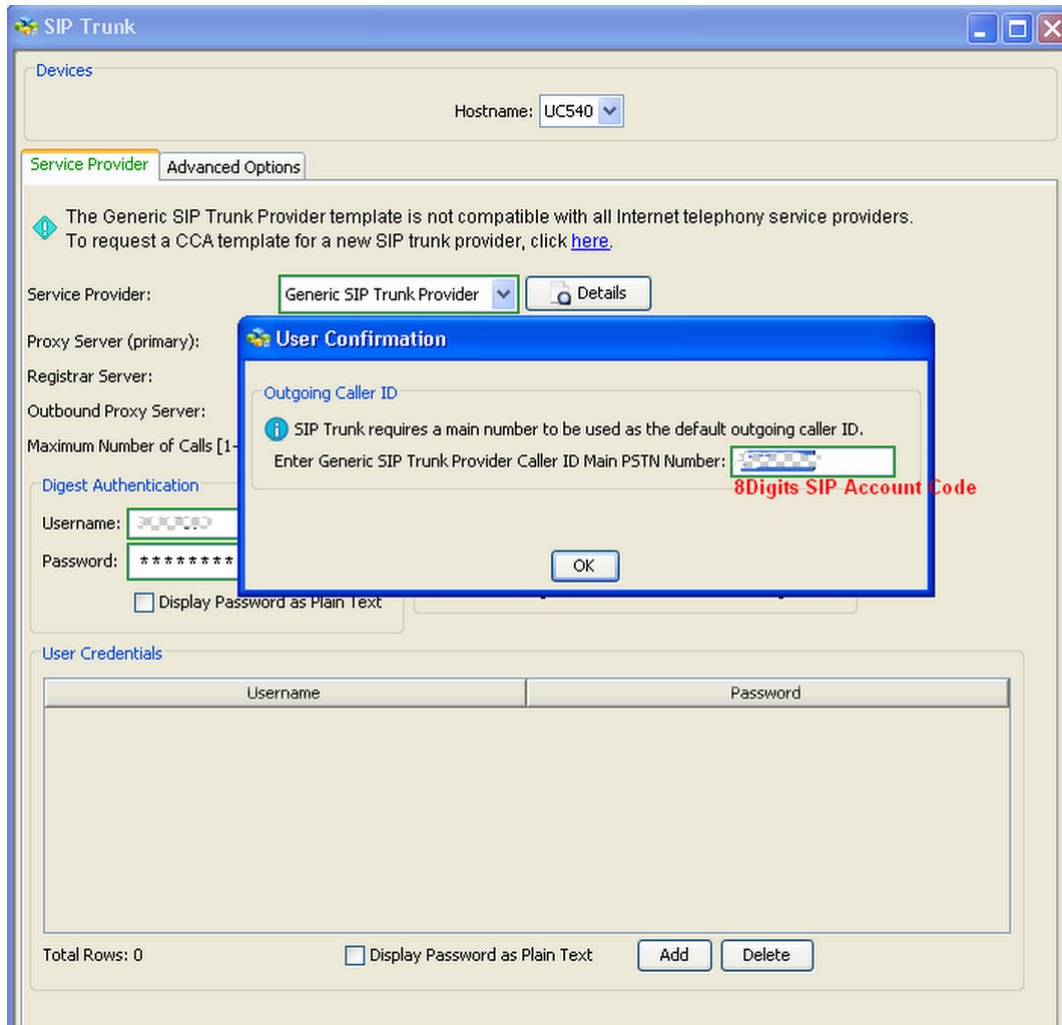
Domain Name Server:

SIP Domain Name : sip17.voicflex.com

DNS Server Address : any valid DNS server (leave blank for the moment)

\* DNS server's: 8.8.8.8 / 8.8.4.4

Once you click the APPLY button, you should see the following screen for default outgoing caller ID. You should add your **8Digits Account Code**



### Modifying source port as 5060:

Once you finished your SIP trunk Settings, you should **save your configuration** and connect to UC540 via command line. (You can connect either console port or telnet its IP address)

By default, Cisco UC540 does not use a static source port; it uses a random unassigned source port. This resulted in outgoing calls not working. Their device was specifically responding with a 404 not found error (which is not very helpful). To get this to work properly, we had to configure the UC540 to use port 5060 for the source port. Basically, you should add "**connection-reuse**" under sip-ua.



```
UC540(config)# sip-ua
UC540(config-sip-ua)# connection-reuse
```

### Modifying registration timeout:

By default, SIP trunks which are configure via CCA has 3600 seconds registration time out value. For changing timeout, you should change the registrar server line which marked as RED.

```
UC540(config)# sip-ua
UC540(config-sip-ua)# no registrar
UC540(config-sip-ua)# registrar dns:sip17.voiceflex.com expires 300
```

### CLI Presentation Settings; sending each subscriber DDI number while making outgoing calls:

Before you start to configure CLI presentation on your PBX, please read information at the following link

<http://wiki.voiceflex.com/How-to-Activate-CLI-feature-on-Voiceflex-Web-Portal.ashx>

According to link above, you need to configure DDI presentation only CLI pass-through scenario,

```
ephone-dn 1 dual-line
 number 504 no-reg both
 label 504
 description Alma Cassin
 name Alma Cassin
!
!
ephone-dn 2 dual-line
 number 505 no-reg both
 label 505
 description Joe Hollingsworth
 name Joe Hollingsworth
```

In this case you should configure voice class sip-profiles to replace name information to DDI number as follows.

The first step is to define the rules. In order to define the rules, use the general command structure given in the previous section. For example:

```
voice class sip-profiles 100
 request INVITE sip-header From modify "\"(Alma Cassin)\\"" "\"02033016010\\""
 request INVITE sip-header From modify "\"(Joe Hollingsworth)\\"" "\"02033016011\\""
```

Based on the configuration above,

Alma Cassin is map to 02033016010 as external DDI number

Joe Hollingsworth is map to 02033016011 as external DDI number

The second step is to apply the rules either to the global or dial-peer level of the UC-540 configuration. In order to apply the rules globally to all calls traversing UC-540, use this command structure:

```
voice service voip
 sip
 sip-profiles 100
```

In order to apply the rules selectively to calls traversing only a particular outgoing dial-peer, use this command structure:

```
dial-peer voice 555 voip
 voice-class sip-profiles 100
```



## Related Information

More information about voice class sip-profiles :

[http://www.cisco.com/en/US/products/sw/voicesw/ps5640/products\\_configuration\\_example09186a0080982499.shtml](http://www.cisco.com/en/US/products/sw/voicesw/ps5640/products_configuration_example09186a0080982499.shtml)

UC500 and SR500 Secure Router Setup

[http://www.cisco.com/en/US/docs/net\\_mgmt/cisco\\_configuration\\_assistant/version1\\_8/quick/guide/English/UC500SR500.html](http://www.cisco.com/en/US/docs/net_mgmt/cisco_configuration_assistant/version1_8/quick/guide/English/UC500SR500.html)

Configuring Multiple Registrars on SIP Trunks

[http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip\\_cg-multi-registrars.html](http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-multi-registrars.html)