

# 4.3 UNENCRYPTED SIP Trunks

Click on the **New Sip Trunk** to go to the Edit Sip Trunk page.

Create SIP Trunk

Name

Unencrypted SIP Trunk

Fallback Group

None

IP / Hostname

unencryptedserver.com

Port

5060

Transport Protocol

UDP

Outbound Proxy

Registrar

Realm Name

test.com

Passcode

\*\*\*\*\*

Virtual Phone Number

001123456789

SRTP Encryption Key

Enable SRTP UAS

☐

SRTP Proxy phone

☐

SRTP Crypto Suite

DEFAULT

Secure RTP

☐

Incoming Call Treatment

New Connected Calls

30

SAI

SD

Direct Media

☒

Send Response to Peer ID

☒

Context

primarysecondary200

Announcement

On

SRTP Signaling Method

RFC6030

Trusted

☐

Create

figure 1. "Edit Sip Trunk" form

In figure 1. "Edit Sip Trunk" form you can read an example configuration for an unencrypted SIP trunk. Mandatory fields for the unencrypted configuration are:

- **NAME:** a meaningful name for this trunk
- **IP / HOSTNAME:** IP address/hostname of the SIP server provided by ITSP
- **PORT:** this is **5060** by RFC
- **TRANSPORT PROTOCOL:** UDP
- **SRTP ENCRYPTION:** disabled (unchecked)

We do also suggest the following values to be set:

- **ANNOUNCEMENT: ON EARLY MEDIA** works fine with the Cisco Unified Communications Manager.

- **DTMF SIGNALING METHOD:** choose your values considering the PBX on the other end of the Trunk. Usually we suggest to choose the value **RF C2833**
- **DIRECT MEDIA:** enabled (checked)
- **SEND REMOTE-PARTY-ID:** enabled (checked)

Other fields in the form depend by your network topology and by the features on the other end PBX.

When you are done with your changes, commit them by clicking on the **Update** icon.

[4.2 ENCRYPTED SIP Trunks ZRTP](#)

[4.4 Authentication in SIP Trunks](#)