



A MITEL
PRODUCT
GUIDE

OpenScape Business

SIP Endpoint configuration

SIP Endpoint connected via Internet

Release Number 12/2024

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History of Changes

Date	Issue	Summary
20.06.2017	3.0	changes for V2R3: new SIP server port, TLS1.2
10.07.2018	3.1	add hint about password requirements
04.12.2024	3.2	editorial changes

Note: The basis for this document is the current OpenScape Business at the time of certification. Since OpenScape Business is constantly developed, input masks and interfaces as well as requirements may change in the future. The settings and entries described here then apply accordingly.

Comments and corrections are welcome, please contact: osbiz-certification@mitel.com

1 Feature description

The feature “**SIP@Home for STUN enabled SIP endpoints**” allows you to register SIP endpoints not only in the local office network, in addition they can register over the Internet.

This document describes the necessary configuration steps to setup connections between SIP-Endpoints and OpenScape Business over the Internet. A typical environment is shown in the following figure:

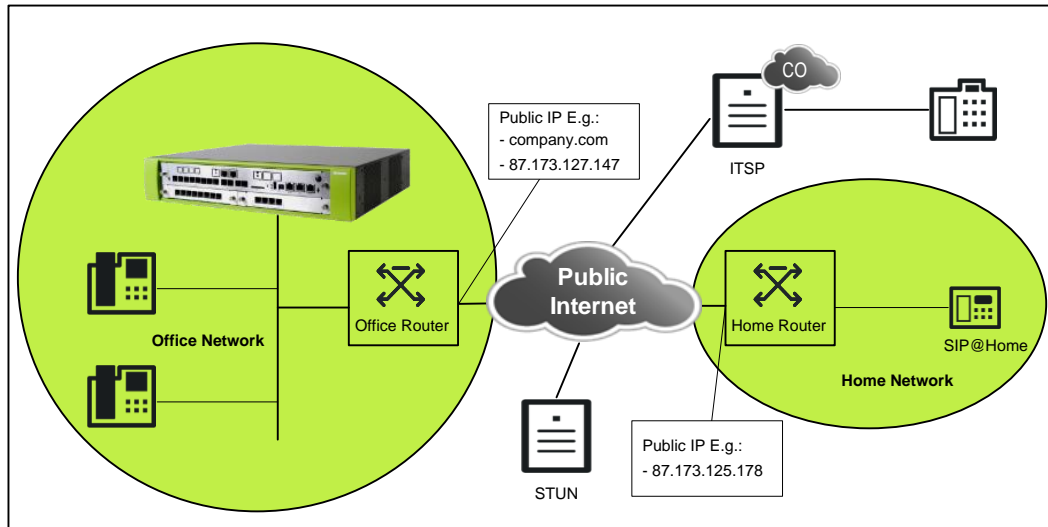


Figure 1 Use case scenario

Connecting SIP endpoints to OpenScape Business requires careful configuration of

- OpenScape Business system
- Office Router
- SIP endpoint

In OpenScape Business you need to allow a SIP endpoint to register over the internet by activating the integrated SBC function for that endpoint (see 2.2). In addition you may activate STUN support if not already used for an ITSP connected to OpenScape Business (see 2.3)

As an endpoint must reach the OpenScape Business system from the Internet you have to configure a port forwarding rule in your office router. (see 3)

Last but not least a SIP endpoint connected over the internet needs appropriate configuration and MUST support STUN (see 4)

In addition to these configuration hints this document provides you with helpful information regarding supported configurations and known limitations.

2 OpenScape Business configuration

2.1 Internet access: supported configurations

There are different possibilities to connect the OpenScape Business system to the internet. The following configurations are supported in a scenario where SIP subscribers shall be able to register via the internet:

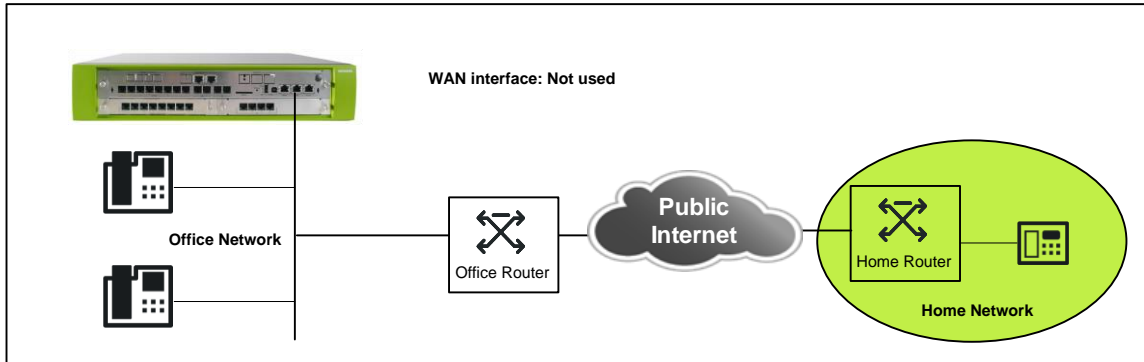


Figure 2 OpenScape Business behind access router connected to LAN2 interface

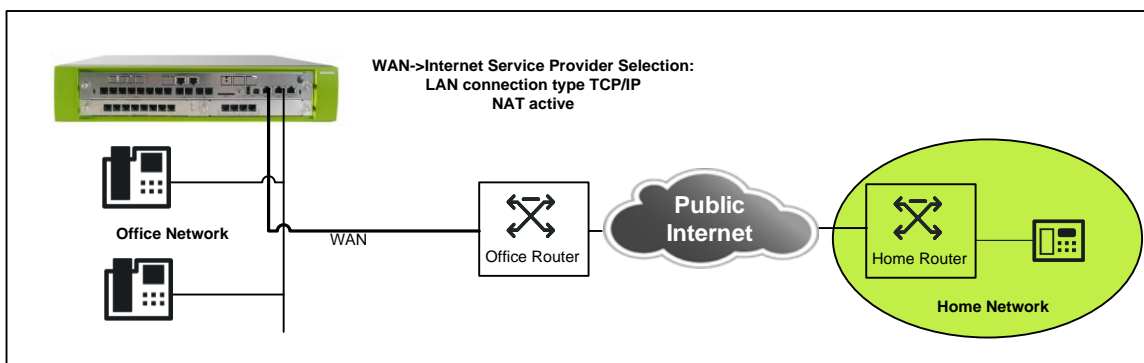
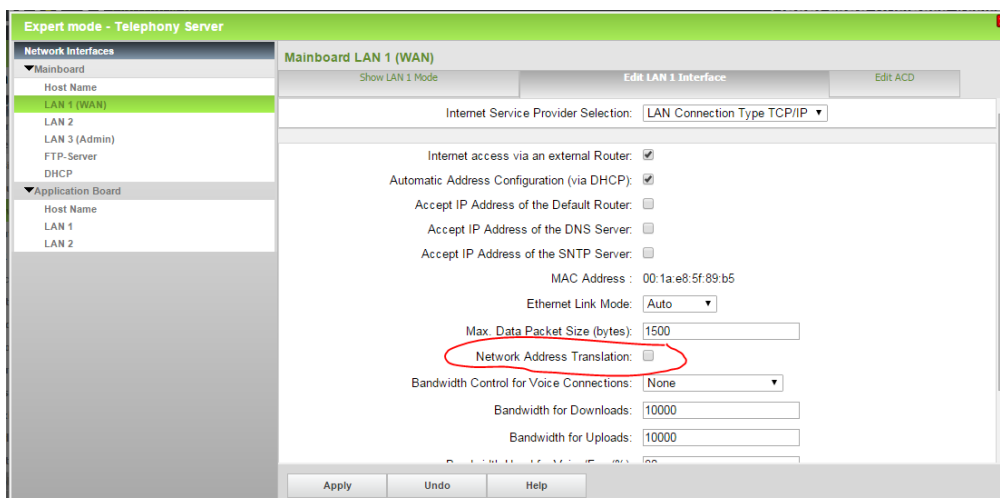


Figure 3 OpenScape Business behind access router connected to LAN1 (WAN) interface



Internet connections using the OpenScape Business as access router behind a modem connected to the WAN interface are NOT supported.

If LAN1 (WAN) is used, the firewall has to be switched off on this interface to allow incoming SIP traffic (Network address translation unchecked)



2.2 Configuring a SIP Endpoint for Internet usage

Please consult the administration manual for a description how to do the basic setup for SIP endpoints.

In addition to that basic configuration the following setting in "Expert Mode" is necessary.

For an endpoint which is allowed to register over the internet the flag "**Internet Registration with internal SBC**" MUST be checked.

Expert mode - Telephony Server

Station

- ▼ Station
 - UP0 Stations
 - ▼ IP Clients
 - System Clients
 - ▼ SIP Clients
 - 60 160 SIP-160
 - 61 161 SIP-161
 - 62 162 SIP-162
 - 63 163 SIP-163
 - 64 164 SIP-164
 - 65 165 SIP-165
 - 66 166 SIP-166
 - 67 100 SIP-167
 - 68 168 SIP-168**
 - 69 169 SIP-169
 - 70 170 SIP-170
 - 71 171 SIP-171
 - 72 172 SIP-172
 - 73 173 SIP-173
 - 74 174 SIP-174
 - 75 175 SIP-175
- RAS User

Station - 68

Type: SIP Client

Call number: 168

Name: SIP-168

Parameter

Authentication active: ☒

Password:

Confirm password:

SIP User ID / Username: SIP-168

Realm: SMO-SIP

Fixed IP address: ☐

IP address: 0.0.0.0

Secondary system ID:

*!Internet Registration with internal SBC: ☒



As the system is accessible from the internet make sure that Authentication is activated and a **STRONG PASSWORD** is used for **ALL** SIP endpoints.

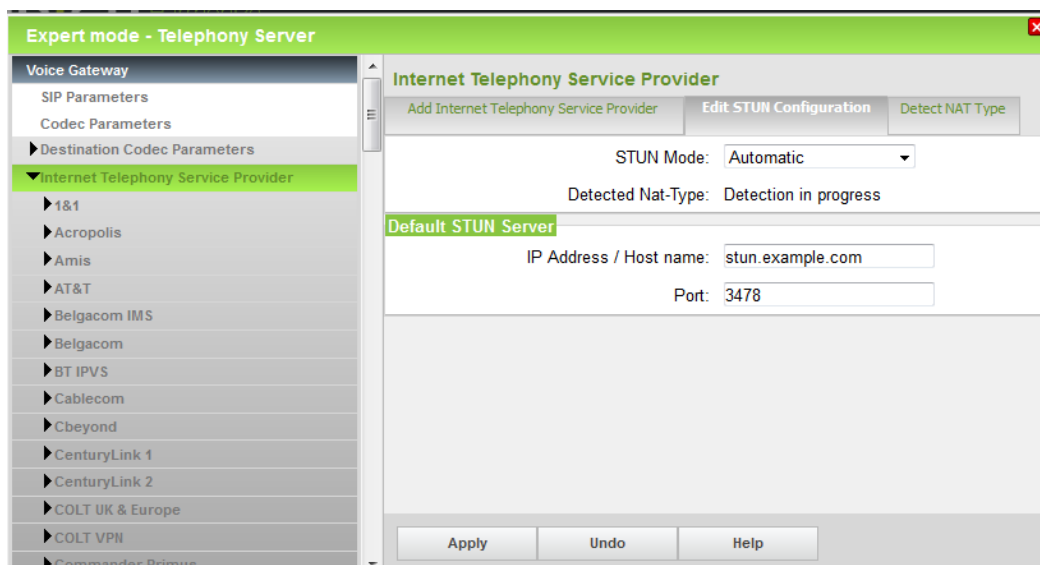
In addition, you may reduce the rights of a SIP station which is allowed to register from the internet e.g. to forbid dialing premium or international numbers.

2.3 Configuring STUN

The integrated SBC function of OpenScape Business must be able to detect its public IP-address and SIP port. This is done using the STUN protocol.

- In case the system is already connected to an ITSP with activated STUN server, no additional configuration is necessary. The system is able to detect its public IP-address and port.
- In case the system is connected to an
 - ITSP with STUN switched off or
 - no ITSP is configured in the system,additional configuration is necessary enabling the system to detect its public IP-address and port.

As shown in the following screenshot, the default STUN server has to be configured in the STUN configuration.



The STUN server configured in "Edit STUN Configuration" will be used only if NO STUN server is configured for an ITSP.

2.4 Port configuration

OpenScape Business provides a dedicated SIP server port for external traffic must be used. No special configuration is needed; the following default ports are used for SIP signalling:

Transport protocol	Default port
UDP / TCP	5070
TLS	5071 (Attention! default TLS port 5061 is used for SIPQ trunking)

In a migrated system these ports will be initialized with the value used in the previous version (default 5060 / 5062) and **MUST** be changed manually in WBM Expert Mode.

2.5 General hints

- **Internet access**

If SIP subscribers register via internet and ITSP connections are used in parallel all VoIP traffic must use the same interface of OpenScape Business. It is NOT allowed to have a configuration where e.g. the ITSP is connected via WAN (using a static route) and public internet access is realized via LAN interface.

- **Holding a remote SIP endpoint:**

Because RTP streams are necessary to keep firewalls open during a call, the integrated SBC must change sendonly media direction attributes to sendrecv media direction when the HOLD feature is invoked by an office phone. As a consequence, the SIP endpoint has no indication to display the hold state.

- **Using TLS transport**

TLS 1.2 transport should be used to connect remote endpoints; therefore the OpenScape Business system must be setup with valid certificates. See appendix for configuration hints.

3 Office Router configuration

3.1 Port forwarding / firewall

In default configuration the firewall in the office router will NOT allow incoming traffic to the OpenScape Business system, thus appropriate port forwarding rules for the SIP port **MUST** be configured in the router.

Transport protocol	Port in system	External port
UDP	5070	5070
TCP	5070	5070 MUST be same as port in system,
TLS	5071	5071 MUST be same as port in system!



NEVER configure a forwarding to the SIP ports used for "internal" traffic (default 5060/5061/5062).

3.2 DynDNS / Internet access with dynamic IP Address

If the Office router is connected to the ISP without a fixed IP-Address, appropriate measures are needed to reach OpenScape Business from the client. This can be achieved by using dynamic DNS. The router must be configured with the DynDNS account and must register the current IP address in regular intervals.



Please note that cost free Dyn-DNS account which expires in regular intervals may lead to temporary malfunction of this feature.

4 SIP Endpoint configuration

The SIP endpoints used for this feature MUST comply with the following requirements:

- Detect own public IP address and port (STUN)
- Use correct public IP:Port in Contact: header field (port determined by for UDP, ephemeral (client) port for TCP)
- Use correct public IP in c: line of SDP
- Use correct public RTP port in m: line of SDP
- Keep NAT bindings active
- Start sending RTP payload
- Use sip: URIs ([tel:-URIs](#) are not supported for this feature)

The following endpoints have been tested and fulfill the above requirements.

4.1 Yealink T19P

Yealink's SIP-T19P entry-level IP phone:

http://yealink.com/product_info.aspx?ProductsCateID=334

Tested SW Version:



The mentioned SW version (**31.72.0.48**) contains important fixes for the use of this feature, other 31.72.x.x versions (including newer versions available on the Yealink download page) cannot be used!

The V7-Unify software is available in the Wiki on the same page where this document is stored.

The functionality will be release by Yealink with V8 in the first quarter of 2015.

Account Configuration:

In this tab all data for the used account and SIP server is entered

Phone Configuration parameter	configured in OpenScope Business: Telephones / Subscribers-> IP Telephones -> Edit
Display Name	Optional, Phone name can only be seen in the network traces, OpenScope Business uses the name configured in system
Register name	SIP User ID / Username
User Name	Call number
Password	Password
Transport	Choose the used transport for your deployment: UDP

Phone Configuration parameter	
NAT	MUST be set to STUN
STUN Server	Enter a reachable STUN server (e.g. stun.sipgate.net) and the STUN port (default 3478).

	A list of public available STUN server is available at e.g. http://www.voip-info.org/wiki/view/STUN
Server Host	Public IP-Address or DNS name of OpenScape Business
Port	SIP Port which is configured at the office router (please note the rules defined for port forwarding in Chap. 3.1)

Yealink | T19 Log Out

Account Status Network DSSKey Features Settings Directory Security

Register

Register Status: Registered

Line Active: Enabled

Label: OsBiz Miami

Display Name: SIP-163

Register Name: SIP-163

User Name: 163

Password:

Enable Outbound Proxy Server: Disabled

Outbound Proxy Server: Port 5060

Transport: UDP

NAT: STUN

STUN Server: stun.sipgate.net Port 3478

SIP Server 1

Server Host: lababel.dyndns.com Port 5070

Server Expires: 300

Server Retry Counts: 3

SIP Server 2

Server Host: Port 5060

Server Expires: 3600

Server Retry Counts: 3

Confirm Cancel

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.



Please note that the T19P phone does not support TLS1.2 and cannot be used with TLS transport.

Hints for troubleshooting:

If a Yealink phone is used, the traces at the remote location can be taken directly from the phones WBM. Start the Pcap Feature and set the system Log Level to 6:

Yealink | T19 Log Out

Settings Status Account Network DSSKey Features Settings Directory Security

Preference

Time & Date

Upgrade

Auto Provision

Configuration

Dial Plan

Voice

Ring

Export or Import Configuration: Durchsuchen... Keine Datei ausgewählt.

Import Export

Pcap Feature: Start Stop Export

Export System Log: Local Server

Export

System Log Level: 6

Confirm Cancel

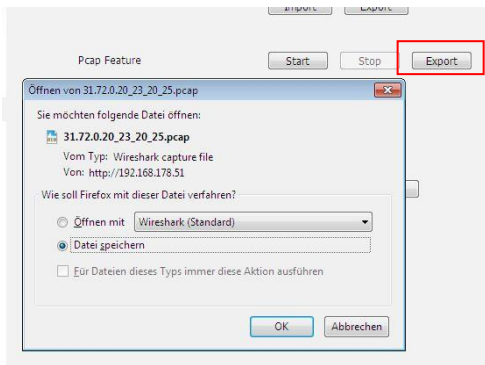
NOTE

Configuration
The configuration parameters for administrator.

Run the scenario where you observed a problem.

At the end "Stop" the Pcap and "Export" it to a file.

In addition "Export" the System log.



4.2 Yealink T41P

Yealink's SIP-T41P feature-rich sip phone for business

http://yealink.com/product_info.aspx?ProductsCateID=313&CateId=147&BaseInfoCateId=313&Cate_Id=313&parentcateid=147

Tested SW Version:



The mentioned SW version (**36.72.0.57**) contains important fixes for the use of this feature, other 36.72.x.x versions (including newer versions available on the Yealink download page) cannot be used!

The V7-Unify software is available in the Wiki on the same page where this document is stored. The functionality will be released by Yealink with V8 in the first quarter of 2015.

Account Configuration:

For T41 the same data have to be entered as for the T19 model.

4.3 Zoiper IOS App

Zoiper is a VoIP softphone that lets you make voice calls with your friends, family, colleagues and business partners.:

<http://www.zoiper.com/en/voip-softphone/download/zoiper3>

Tested SW Version: 2.17

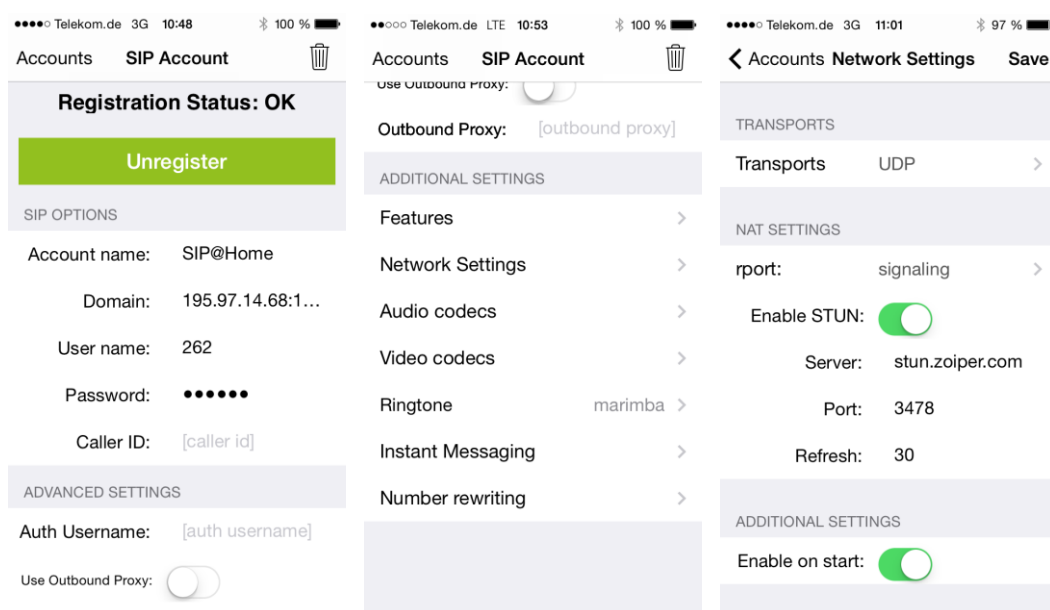
Account Configuration:

Go to Accounts: Create Account -> SIP account

In this tab all data for the used account and SIP server is entered



Phone Configuration parameter	configured in OpenScape Business: Telephones / Subscribers-> IP Telephones -> Edit
SIP OPTIONS	
Account Name	Optional, Phone name can only be seen in the network traces, OpenScape Business uses the name configured in system
Domain	Public IP-Address or DNS name of OpenScape Business and SIP Port which is configured at the office router (please note the rules defined for port forwarding in Chap. 3.1)
User Name	Call number
Password	Password
Caller ID	
ADVANCED SETTINGS	
Auth Username	SIP User ID / Username
ADDITIONAL SETTINGS -> Network settings	
Transport	UDP is offered in default, keep this setting
Enable STUN	MUST be activated



4.4 Zoiper Android App

Zoiper is a VoIP softphone that lets you make voice calls with your friends, family, colleagues and business partners.:

https://play.google.com/store/apps/details?id=com.zoiper.android.app&referrer=utm_source%3Dzoiper.com

Tested SW Version: 1.51.17

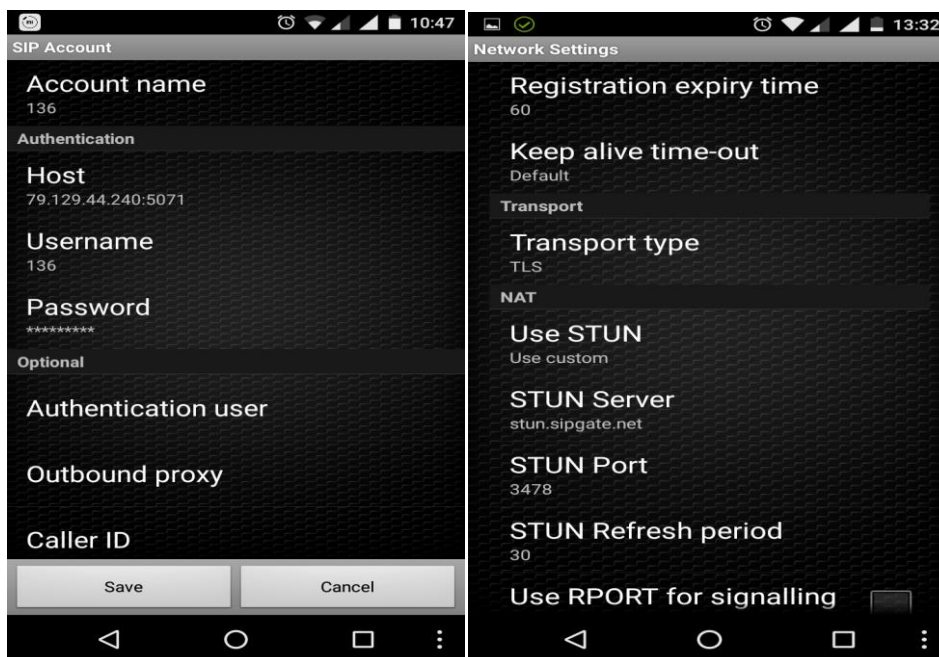
Android version: 6.0.1 (Marshmallow)

Account Configuration:

Go to Accounts: Create Account -> SIP account

In this tab all data for the used account and SIP server is entered

Phone Configuration parameter	configured in OpenScape Business: Telephones / Subscribers-> IP Telephones -> Edit
SIP Account	
Account Name	Optional, Phone name can only be seen in the network traces, OpenScape Business uses the name configured in system
Host	Public IP-Address or DNS name of OpenScape Business and SIP Port (SIP_TLS_SUB_EXT) which is configured at the office router (please note the rules defined for port forwarding in Chap. 3.1).
Username	Call number
Password	Password
Optional settings	
Authentication user	SIP User ID / Username
Optional settings -> network settings	
Transport type	UDP or TLS
Enable STUN	MUST be activated



5 Home-/SOHO-Router

No specific configuration is necessary for this feature in the Home router.

The Home router used for this feature **MUST** comply with the following requirements:

- The Home router **MUST** provide VoIP enabled NAT (no symmetric NAT),
- ALG function in the router **MUST** be deactivated if present.

Please make sure that there is sufficient bandwidth available for real time traffic at the remote location. This needs to be taken into account when e.g. using asymmetric DSL connections, which may have reduced upload speed.

6 Known restrictions and limitations

6.1 Use of video is not possible

The implementation of the integrated SBC-light allows for a single media stream per session. It does NOT allow using more than 1 media stream (e.g. voice and video).

6.2 Use of endpoints without STUN support is not possible

The implementation of the integrated SBC-light relies on correct signaling information in terms of SIP signaling and media addresses. It does NOT allow the connection of phones which do not provide correct public IP address information when connected behind a router (e.g. phones without STUN capability).

6.3 Use of secured connections

TLS1.0 is not longer supported.

TLS connections for SIP subscribers are supported at the LAN interface of the OpenScape Business system only.

TLS at the WAN interface is NOT supported.

SRTP payload using SDES signalling is NOT supported

7 Capacities

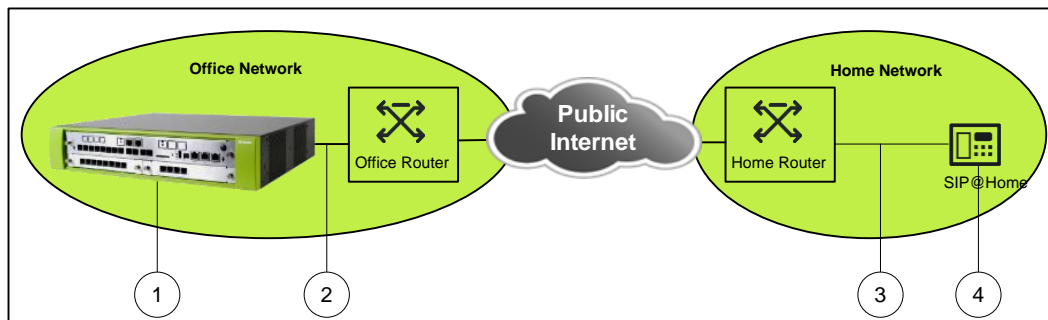
For all "remote VoIP" scenarios a shared pool of "RTP proxy" resources is used in OpenScape Business. This includes:

- active ITSP calls (1 RTP proxy channel per call),
- active SIP@Home calls (1 RTP proxy channel per device),
- active HFA@Home calls (1 RTP proxy channel per device),
- active myPortal to go VoIP @Home calls (1 RTP proxy channel per client),
- Circuit (Cloud) calls (1 RTP proxy channel per call)

The following resources are available:

System variant	RTP proxy channels
OpenScape Business X1/X3/X5/X8 with or without Booster card/server	60
OpenScape Business S	180

8 Troubleshooting



As this feature connects several networks for a connection, in case of connection problems the following traces are needed:

1. Internal trace from OpenScape Business with the following Trace profiles activated:
 - Voice_Fax_connection
 - SIP_Interconnection_Subscriber_ITSP
 - SIP_Registration
2. Wireshark trace capturing the traffic between the office router and the OpenScape Business system. This could be a TCP-dump from the router or a capture taken from the LAN
3. Wireshark trace from the remote location capturing the traffic between the affected SIP phone and the Home-/SOHO-Router. This could be a TCP-dump from the router or a capture taken from the LAN
4. When available diagnostic logs/trace from the device at remote location
5. Information about Setup, e.g.
 - Used device (type and software release) at remote location
 - Used router at remote location
 - Used router at office location
 - List of IP addresses of all involved entities (phone, routers, OSBiz system)

9 Appendix

9.1 Configuration for use of TLS

9.1.1 Security configuration

The system flag "*SPE support*" has to be activated under Basic settings -> System -> System Flags

Expert mode - Telephony Server

Basic Settings

- System
 - System Flags
 - Time Parameters
 - Display
 - DISA
 - Intercept/Attendant/Hotline
 - LDAP
 - Texts
 - Flexible menu
 - Speed Dials
 - Service Codes
 - Gateway
- DynDNS
- Quality of Service
- Date and Time
- Port Management
- Call Charges
 - Voicemail / Announcement Player
 - Phone Parameter Deployment
 - Power Management

System Flags

Edit System Flags

Automatic redial: ☐

Voice mail Node call number: ☐

Call Pickup after automatic recall: ☐

Configurable CLIP: ☒

Caller list at destination in case of Forward Line: ☐

Call forwarding after deflect call / single step transfer: ☐

Follow call management in case of deflect call / single step transfer: ☐

Warning tone during voice recording: ☒

Calling number in pick-up groups / ringing groups / CFN /RNA: ☒

SPE support: ☒

SPE advisory tone: ☒

SIP Prov. to SIP Prov. transit: ☐

Transparent dialing of * and # on trunk interfaces: ☐

Add seizure code for MEX: ☐

CMI MWI Ringer: ☐

Open numbering scheme

active: ☐

Node callnumber:

Transit permission

Feature transit: ☒

Apply Undo Help

9.1.2 Certificate generation

For SIP devices OpenScope Business acts as a TLS server and thus needs to have a TLS server certificate. You may install your own certificate if available (e.g. provided by ITSP) or create a new one. The following steps describe how to create a self-signed server certificate.



With the current implementation only one certificate can be installed for all SIP interfaces (ITSP, SIPQ-interconnection, SIPQ-trunk, SIP subscriber).

First a "CA Certificate" has to be generated. Navigate to Security->SSL-> Certificate Generation and open the "Generate a CA Certificate" page. **Enter the corresponding data and apply the changes. As a result the certificate is stored on the OpenScope Business system.**

Expert mode - Telephony Server

Security

- Application Firewall
- Deployment and Licensing Client (DLSC)
- Signaling and Payload Encryption (SPE)
- VPN
- SSL
 - Certificate Generation
 - Certificate Management

Display General Information

Generate CA Certificate | Generate Self-Signed Certificate

Name of the Certificate: SPE Certificate

Serial Number of Certificate: 1

Type of Signature Algorithm: sha1RSA

Public Key Length: 1536

Start Time of Validity Period (GMT)

Day	Month	Year
19	9	2014
Hour	Min.	Sec.
0	0	0

End Time of Validity Period (GMT)

Day	Month	Year
19	9	2024
Hour	Min.	Sec.
0	0	0

Subject Name

Country (C): DE

Organization (O): UNIFY GmbH & Co. KG

Organization Unit (OU): PH HQ DPL

Common Name (CN): SPE CA

Subject Alternative Name

Distinguished Name Format: Other Format

Subject Alternative Name Extension: Email address (optional)

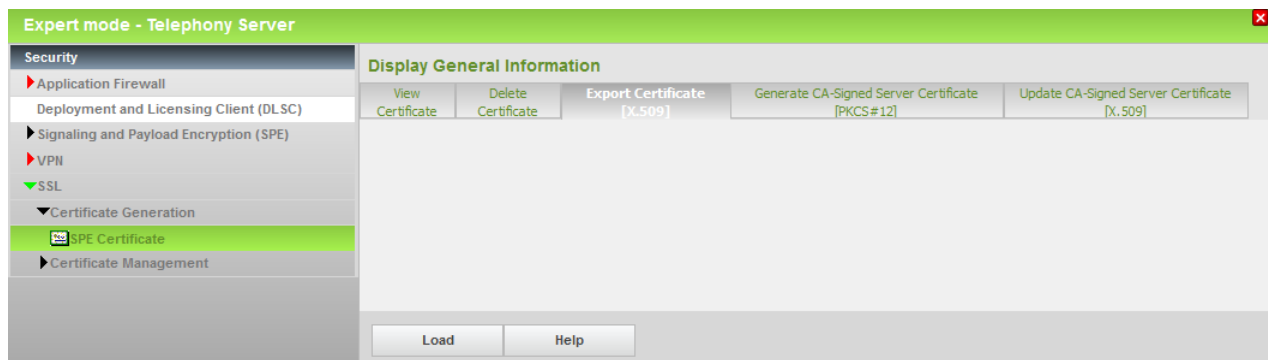
Subject Alternative Name: (optional)

CRL Distribution Point Type: DNS Name (optional)

CRL Distribution Point: (optional)

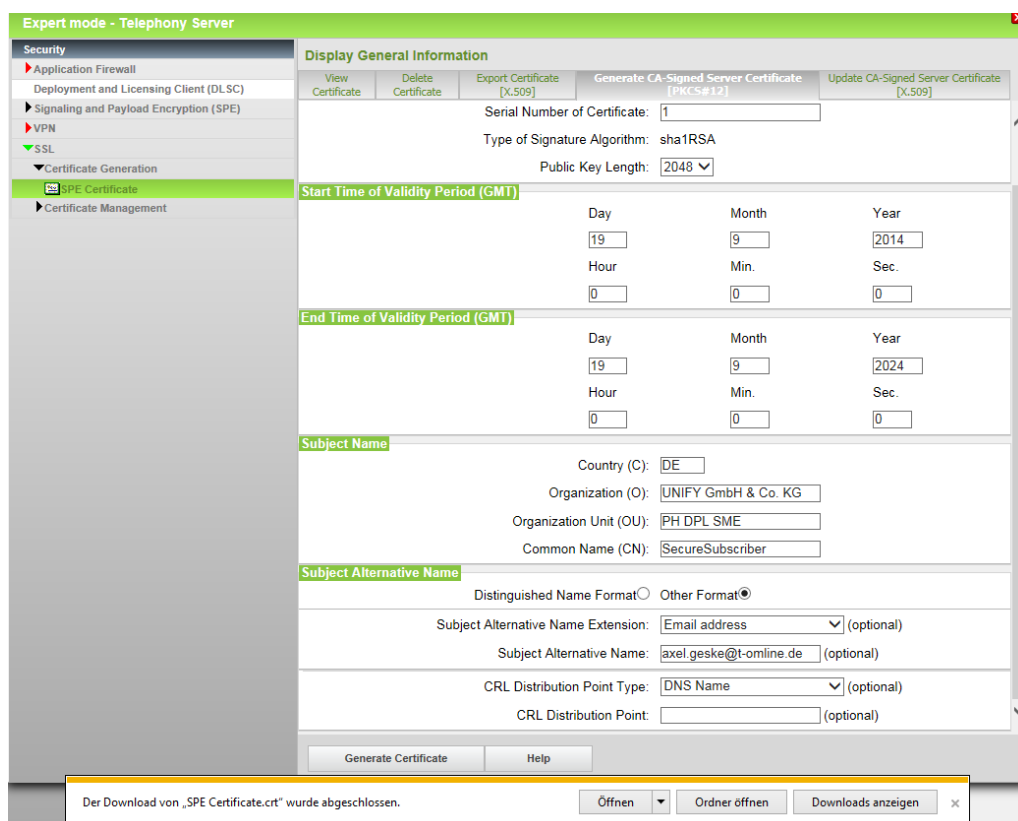
Apply Help

In the next step the "CA Certificate" has to be exported in X.509 format



Choose and appropriate place on your computer to store the CA certificate (default name: Common Name.crt).

Now the CA signed server certificate can be generated and exported in PKCS#12 format.



The server certificate is stored as BasedOn"Common Name".p12

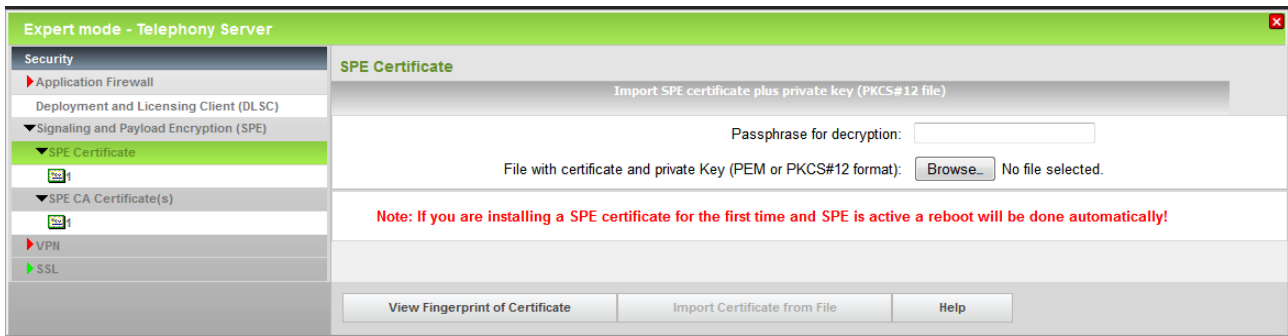
After this step two files are present: :

1. CA certificate "Common Name".crt
2. PKCS#12 Certificate BasedOn"Common Name".p12

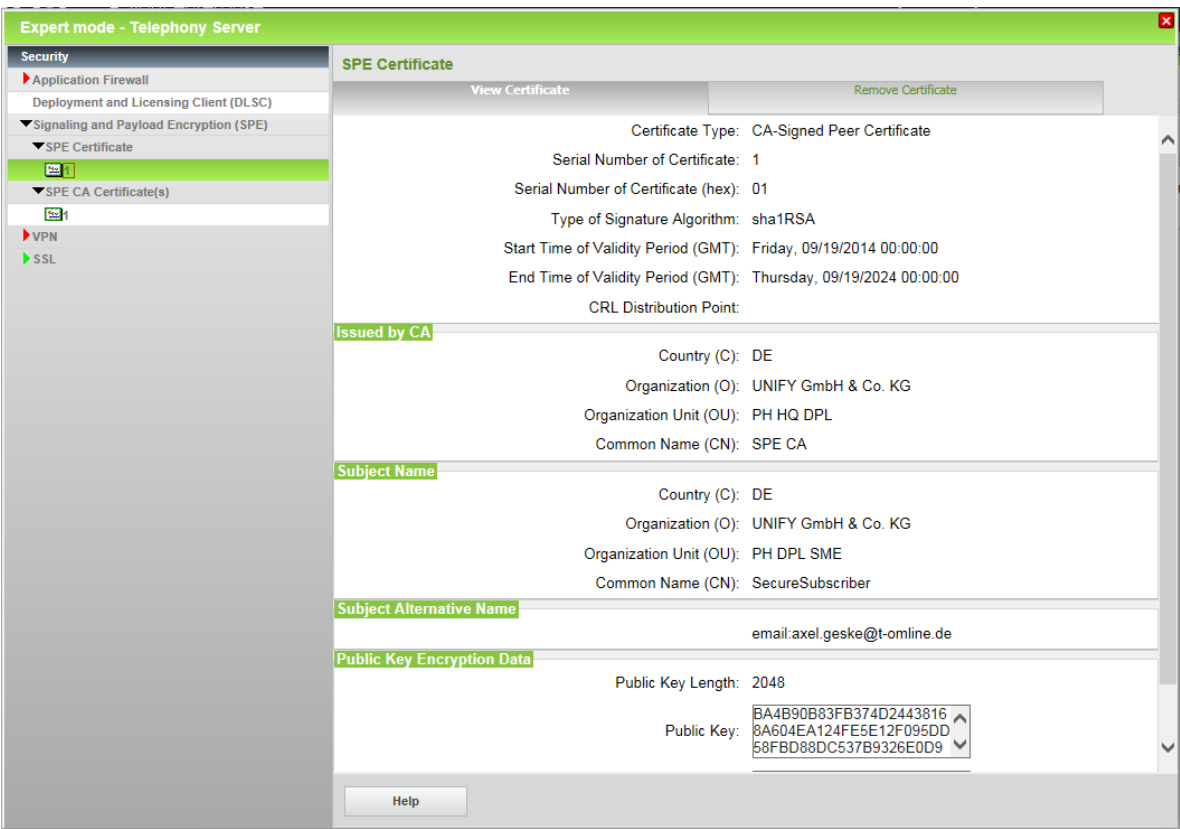
With these certificates the OpenScape Business system can be configured to act as a TLS server.

9.1.3 Certificate configuration

The Certificate and CA Certificate generated as described in the previous chapter must be installed on the OpenScope Business system (in Signalling and Payload encryption (SPE) section).



The installed Certificates can be viewed:



Expert mode - Telephony Server

Security
Application Firewall
Deployment and Licensing Client (DLSC)
▼ Signaling and Payload Encryption (SPE)
▼ SPE Certificate
1
▼ SPE CA Certificate(s)
VPN
SSL

SPE CA Certificate(s)
View Certificate
Display CRL
Remove Certificate

Certificate Type: Self-Signed CA Certificate
Serial Number of Certificate: 1
Serial Number of Certificate (hex): 01
Type of Signature Algorithm: sha1RSA
Start Time of Validity Period (GMT): Friday, 09/19/2014 00:00:00
End Time of Validity Period (GMT): Thursday, 09/19/2024 00:00:00
CRL Distribution Point:

Issued by CA
Country (C): DE
Organization (O): UNIFY GmbH & Co. KG
Organization Unit (OU): PH HQ DPL
Common Name (CN): SPE CA

Subject Name
Country (C): DE
Organization (O): UNIFY GmbH & Co. KG
Organization Unit (OU): PH HQ DPL
Common Name (CN): SPE CA

Subject Alternative Name

Public Key Encryption Data
Public Key Length: 1536
Public Key:
D69723FCAECCCF39DC3C
6B17E71218D0315019E84E
1E0048DAEFEDE68149009

Help

9.2 Technical Background

The following chapter should give some details about the technical background about this feature. This information might be useful in case you need to do troubleshooting.

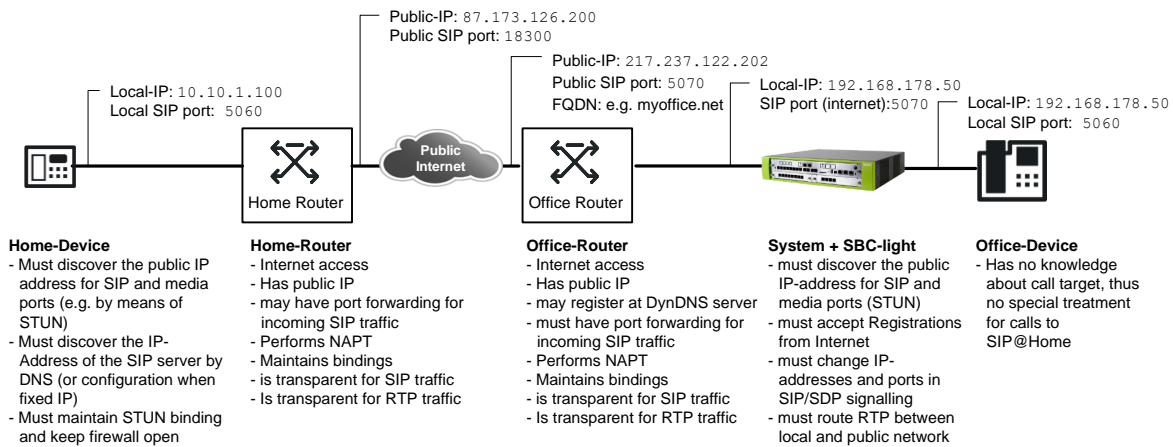


Fig 8.2-1: SIP@Home Scenario overview

For successful registration the SIP@Home phone must determine its public IP-address and port for SIP.

In addition, the phone is configured with the FQDN or public IP address and port of the OpenScope Business Server.

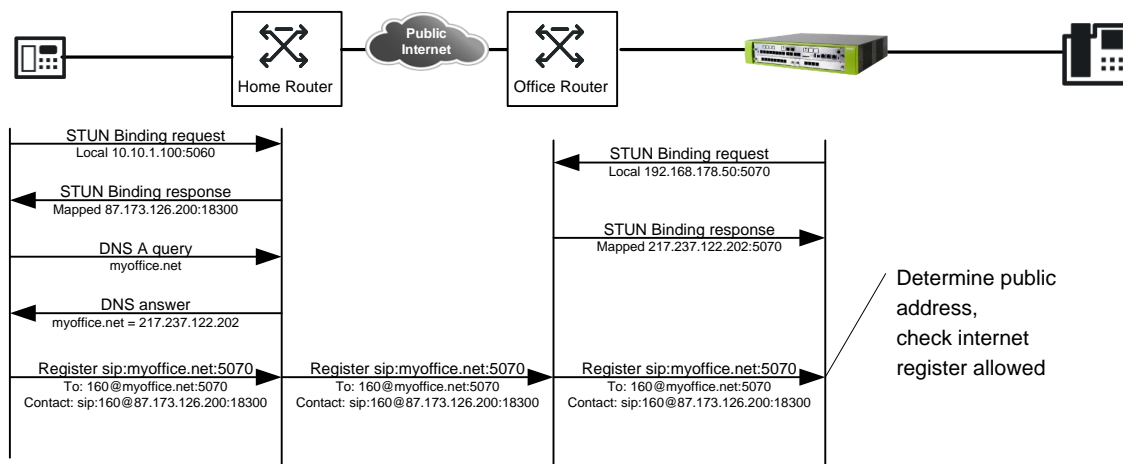


Fig 8.2-2: Detection of public SIP address

The SIP@Home registers with the OpenScope Business system where the information how to reach the phone is retrieved from the received message and stored for later use.

In case of using UDP transport the information is taken from the contact header field.

In case of TLS the information available in the contact header field is not sufficient (as the port cannot be determined by STUN), thus the system will use the transport address from where the packet is received.

The OpenScope Business Server use STUN as well to determine its public IP address and port. The SIP port is checked in regular intervals (STUN monitor function, every 15 seconds). The STUN binding for RTP is done whenever a call is established.

The following figure shows an example for call establishment and the impact of STUN for media transport:

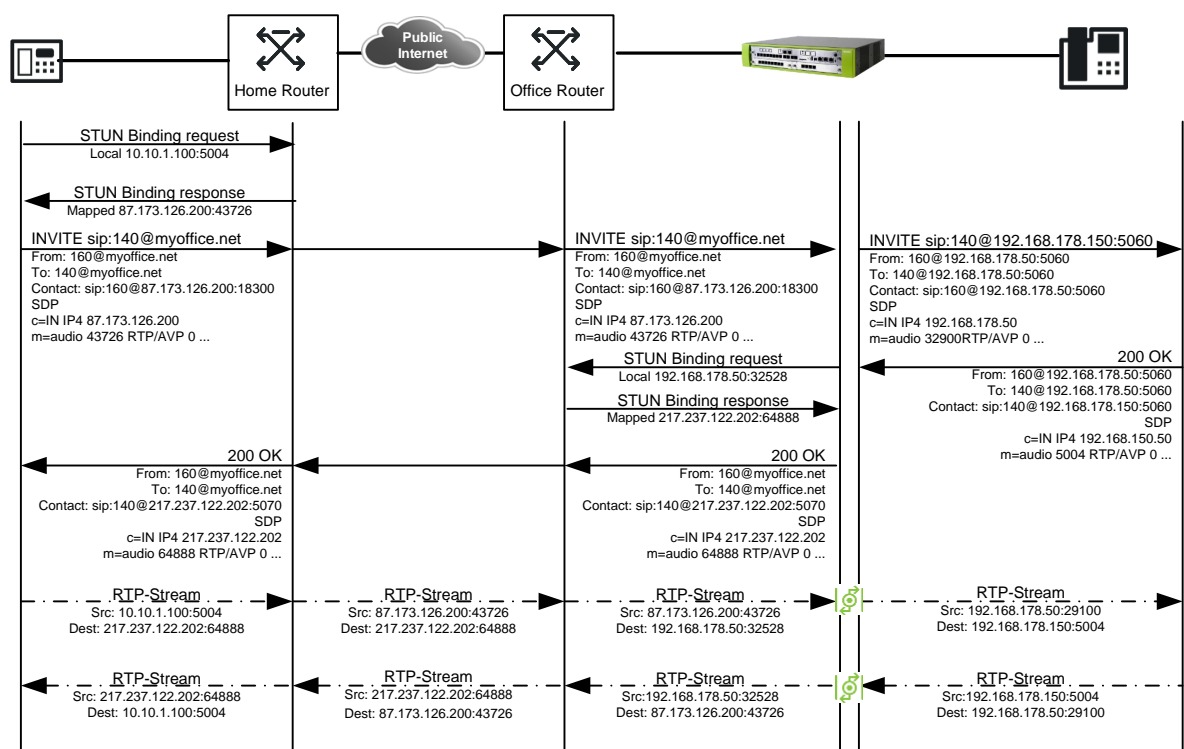


Fig 8.2-3: Call setup and detection of public RTP address

STUN has to be finished before the system can send the SDP information towards the SIP endpoint.

