estos uaCSTA Server for SIP Phones

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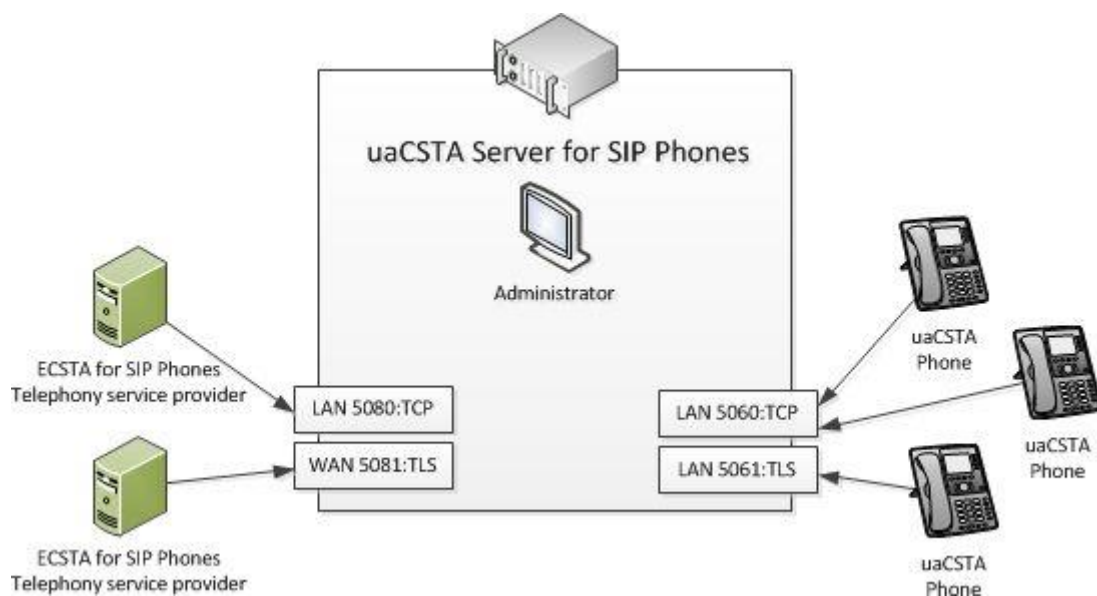
1   estos uaCSTA Server for SIP Phones

estos uaCSTA Server for SIP Phones is a CSTA middleware for SIP telephones capable of uaCSTA. The solution consists of uaCSTA Server for SIP Phones and ECSTA for SIP Phones. They permit control and monitoring of telephones from applications compatible with Microsoft® TAPI. uaCSTA Server for SIP Phones provides controllable telephone resources to registered clients. uaCSTA Server for SIP Phones is thereby the central binding element between all participating components. ECSTA for SIP Phones is a telephony service provider for Tapi 2.1 (and 2.2 & 3.0).

Conceptually, the uaCSTA Server for SIP Phones is a registrar and a proxy. All components with would like to exchange telephony information must log on to the registrar. As a proxy, it can transfer information given a separated network infrastructure (its own network for telephony, VLAN or physically separated). The default SIP and CSTA protocols as well as through certificate-based TLS, permit secure standardized communications between all participating components.

As a telephony service provider (TSP), ECSTA for SIP Phones provides a TAPI conforming to standards. The driver will be used in connection with applications using TAPI (UCServer, ProCall and other TAPI applications).

uaCSTA Server for SIP Phones and ECSTA for SIP Phones are provided as an MSI installer and can be used in the current 32 and 64 bit Windows® operating systems. Both components can be installed on one and the same machine.

2 Requirements
2.1 Installation and Operation

uaCSTA Server for SIP Phones and ECSTA for SIP Phones can be used on 32- & 64-bit Windows® operating systems. Both products can be installed on the same computer. If other SIP applications have been installed on the computer on which the server should be installed, the usage of the default SIP and SIPTLS ports (5060 & 5061) must be checked. If these ports are already in use, you will have to use other ports available.

Supported Operating Systems:

- Windows® 10 64bit
- Windows® 11 64bit
- Windows Server® 2016
- Windows Server® 2019
- Windows Server® 2022

2.2 Communication with Telephones

In order that uaCSTA Server for SIP Phones can communicate with the telephones, they will have to be registered with the server. To do this, a SIP identity, through which the telephone will be registered with the server, must be configured on each telephone to be controlled. These settings can be stored in the telephone through the Provisioning dialog integrated into the Administrator. As soon as the telephone registers itself with the server, it can be monitored and controlled. Environments in which the telephones are found in a VLAN or physically separated network are supported. To do this, the server must be integrated into this network using a network interface. You will find detailed information about the configuration of the IP interface and the login information in the settings for the Telephone Connection.

2.3 Communications with ECSTA for SIP Phones

ECSTA for SIP Phones will contact the server analogously to the telephones. Here too, the client must be able to contact the server. You will find detailed information about the configuration of the IP interfaces and the login information in the settings for the Client Connection.

2.4 Supported Telephones and Features

Auerswald
COMfortel 1400 IP, COMfortel 2600 IP, COMfortel 3600 IP
Auerswald: From Firmware Version 2.8 A

Supported CSTA Features (associative events are supported correspondingly):

CSTA Feature	Description	supported
MakeCall	Makes phone calls	yes
AnswerCall	Answer call	yes
ClearConnection	Terminate phone call	yes
HoldCall	Hold call	yes

RetrieveCall	Retrieve phone call on hold	yes
AlternateCall	Switch between the active phone call and the call on hold	Not supported, but emulated using the HoldCall & RetrieveCall functions.
ConferenceCall	Combine the active phone call with the call on hold to create a conference call.	yes
TransferCall	Connects the active phone call and the call on hold.	yes
GenerateDigits	Dials DTMF tones.	yes
DialDigits	Dials the digital tones for a phone call.	Not supported
ConsultationCall	Initiates a callback call.	yes
SingleStepTransferCall	Transfers an active call blindly	yes
DeflectCall	Redirects calls	yes
DirectedPickupCall	Picks up a call from another participant	Not supported
GetDoNotDisturb	Checks the Do Not Disturb indicator.	yes
SetDoNotDisturb	Sets the Do Not Disturb indicator.	yes
GetForwarding	Checks for forwarded phone numbers.	yes
SetForwarding	Sets up call forwarding.	yes
MonitorStart	Starts monitoring the telephone.	yes
MonitorStop	Stops monitoring the telephone.	yes
GetSwitchingFunctionDevices	Checks the configured identities.	yes
SnapShotDevice	Consistency check	yes
SnapShotCall	Consistency check	Not supported

SystemRegister	Subscribes to notifications of configuration changes.	yes
SystemRegisterCancel	Unsubscribes from notifications for configuration changes.	yes
RequestSystemStatus	Checks the telephone status.	yes

Mitel
6863i, 6865i, 6867i, 6869i
Mitel: From FirmwareVersion 4.1.0 Service Pack 2

Supported CSTA Features (associative events are supported correspondingly):

CSTA Feature	Description	supported
MakeCall	Makes phone calls	yes
AnswerCall	Answer call	yes
ClearConnection	Terminate phone call	yes
HoldCall	Hold call	yes
RetrieveCall	Retrieve phone call on hold	yes
AlternateCall	Switch between the active phone call and the call on hold	Not supported, but emulated using the HoldCall & RetrieveCall functions.
ConferenceCall	Combine the active phone call with the call on hold to create a conference call.	yes
TransferCall	Connects the active phone call and the call on hold.	yes
GenerateDigits	Dials DTMF tones.	yes
DialDigits	Dials the digital tones for a phone call.	Not supported
ConsultationCall	Initiates a callback call.	yes
SingleStepTransferCall	Transfers an active call blindly	yes

DeflectCall	Redirects calls	yes
DirectedPickupCall	Picks up a call from another participant	Not supported
GetDoNotDisturb	Checks the Do Not Disturb indicator.	yes
SetDoNotDisturb	Sets the Do Not Disturb indicator.	yes
GetForwarding	Checks for forwarded phone numbers.	yes
SetForwarding	Sets up call forwarding.	yes
MonitorStart	Starts monitoring the telephone.	yes
MonitorStop	Stops monitoring the telephone.	yes
GetSwitchingFunctionDevices	Checks the configured identities.	yes
SnapShotDevice	Consistency check	yes
SnapShotCall	Consistency check	yes
SystemRegister	Subscribes to notifications of configuration changes.	Not supported
SystemRegisterCancel	Unsubscribes from notifications for configuration changes.	Not supported
RequestSystemStatus	Checks the telephone status.	yes

snom
D3xx Serie, D7xx Serie, 300, 320, 360, 370, 720, 760, 820, 821, 870, D120* (*For D120 No Auto Provisioning)
snom: From Firmware Version 8.7.5.13

Supported CSTA Features (associative events are supported correspondingly):

CSTA Feature	Description	supported
MakeCall	Makes phone calls	yes
AnswerCall	Answer call	yes
ClearConnection	Terminate phone call	yes

HoldCall	Hold call	yes
RetrieveCall	Retrieve phone call on hold	yes
AlternateCall	Switch between the active phone call and the call on hold	Not supported, but emulated using the HoldCall & RetrieveCall functions.
ConferenceCall	Combine the active phone call with the call on hold to create a conference call.	yes
TransferCall	Connects the active phone call and the call on hold.	yes
GenerateDigits	Dials DTMF tones.	yes
DialDigits	Dials the digital tones for a phone call.	Not supported
ConsultationCall	Initiates a callback call.	yes
SingleStepTransferCall	Transfers an active call blindly	yes
DeflectCall	Redirects calls	yes
DirectedPickupCall	Picks up a call from another participant	Not supported
GetDoNotDisturb	Checks the Do Not Disturb indicator.	yes
SetDoNotDisturb	Sets the Do Not Disturb indicator.	yes
GetForwarding	Checks for forwarded phone numbers.	yes
SetForwarding	Sets up call forwarding.	yes
MonitorStart	Starts monitoring the telephone.	yes
MonitorStop	Stops monitoring the telephone.	yes
GetSwitchingFunctionDevices	Checks the configured identities.	yes
SnapShotDevice	Consistency check	yes

SnapShotCall	Consistency check	Not supported
SystemRegister	Subscribes to notifications of configuration changes.	yes
SystemRegisterCancel	Unsubscribes from notifications for configuration changes.	yes
RequestSystemStatus	Checks the telephone status.	yes

Yealink
VP59, SIP-T58V, SIP-T58A, SIP-T56A, T57W, T54W, T53W, T53, SIP-T54S, SIP-T52S, SIP VP-T49G, T4S, SIP-T48G, SIP-T46G, SIP-T42G, SIP-T41P, SIP-T40G, SIP-T40P, SIP-T29G, SIP-T27G, SIP-T23G, SIP-T23P, SIP-T27P
Yealink: From Firmware Version x.81.0.15

Supported CSTA Features (associative events are supported correspondingly):

CSTA Feature	Description	supported
MakeCall	Makes phone calls	yes
AnswerCall	Answer call	yes
ClearConnection	Terminate phone call	yes
HoldCall	Hold call	yes
RetrieveCall	Retrieve phone call on hold	yes
AlternateCall	Switch between the active phone call and the call on hold	Not supported, but emulated using the HoldCall & RetrieveCall functions.
ConferenceCall	Combine the active phone call with the call on hold to create a conference call.	yes
TransferCall	Connects the active phone call and the call on hold.	yes
GenerateDigits	Dials DTMF tones.	yes
DialDigits	Dials the digital tones for a phone call.	Not supported

ConsultationCall	Initiates a callback call.	yes
SingleStepTransferCall	Transfers an active call blindly	yes
DeflectCall	Redirects calls	yes
DirectedPickupCall	Picks up a call from another participant	Not supported
GetDoNotDisturb	Checks the Do Not Disturb indicator.	yes
SetDoNotDisturb	Sets the Do Not Disturb indicator.	yes
GetForwarding	Checks for forwarded phone numbers.	yes
SetForwarding	Sets up call forwarding.	yes
MonitorStart	Starts monitoring the telephone.	yes
MonitorStop	Stops monitoring the telephone.	yes
GetSwitchingFunctionDevices	Checks the configured identities.	yes
SnapShotDevice	Consistency check	yes
SnapShotCall	Consistency check	Not supported
SystemRegister	Subscribes to notifications of configuration changes.	Not supported
SystemRegisterCancel	Unsubscribes from notifications for configuration changes.	Not supported
RequestSystemStatus	Checks the telephone status.	ja

Polycom
VVX 201, VVX 301/311, VVX 401/411, VVX 501, VVX 601, VVX x50-Serie, CCX-Serie
Polycom: From Firmware Version 6.1.0.6189

Supported CSTA Features (associative events are supported correspondingly):

CSTA Feature	Description	supported
MakeCall	Makes phone calls	yes

AnswerCall	Answer call	yes
ClearConnection	Terminate phone call	yes
HoldCall	Hold call	yes
RetrieveCall	Retrieve phone call on hold	yes
AlternateCall	Switch between the active phone call and the call on hold	Not supported, but emulated using the HoldCall & RetrieveCall functions.
ConferenceCall	Combine the active phone call with the call on hold to create a conference call.	yes
TransferCall	Connects the active phone call and the call on hold.	yes
GenerateDigits	Dials DTMF tones.	Not supported
DialDigits	Dials the digital tones for a phone call.	Not supported
ConsultationCall	Setup transfer	yes
SingleStepTransferCall	Transfers an active call blindly	yes
DeflectCall	Redirects calls	yes
DirectedPickupCall	Picks up a call from another participant	Not supported
GetDoNotDisturb	Checks the Do Not Disturb indicator.	Not supported
SetDoNotDisturb	Sets the Do Not Disturb indicator.	Not supported
GetForwarding	Checks for forwarded phone numbers.	yes
SetForwarding	Sets up call forwarding.	yes
MonitorStart	Starts monitoring the telephone.	yes
MonitorStop	Stops monitoring the telephone.	yes

GetSwitchingFunctionDevices	Checks the configured identities.	yes
SnapShotDevice	Consistency check	yes
SystemRegister	Subscribes to notifications of configuration changes.	Not supported
SystemRegisterCancel	Unsubscribes from notifications for configuration changes.	Not supported
RequestSystemStatus	Checks the telephone status.	yes

Gigaset
Base station: N870 IP PRO; Devices: R650H PRO und SL750H PRO
Gigaset Base station: Firmware Version 2.39.0

The Gigaset base station only provides a very limited scope of CSTA functions. Supported CSTA functions:

CSTA Function	Description	Support
MakeCall	Makes phone calls	yes
AnswerCall	Answer call	yes
ClearConnection	Terminate phone call	Either from the connection status or while the connection is being established, but then only on the part of the calling subscriber.
HoldCall	Hold call	Not supported
RetrieveCall	Retrieve phone call on hold	Not supported
AlternateCall	Switch between the active phone call and the call on hold	Not supported
ConferenceCall	Combine the active phone call with the call on hold to create a conference call.	Not supported
TransferCall	Connects the active phone call and the call on hold.	Not supported
GenerateDigits	Dials DTMF tones.	Not supported

DialDigits	Dials the digital tones for a phone call.	Not supported
ConsultationCall	Setup transfer	Not supported
SingleStepTransferCall	Transfers an active call blindly	Not supported
DeflectCall	Redirects calls	Not supported
DirectedPickupCall	Picks up a call from another participant	Not supported
GetDoNotDisturb	Checks the Do Not Disturb indicator.	yes
SetDoNotDisturb	Sets the Do Not Disturb indicator.	yes
GetForwarding	Checks for forwarded phone numbers.	Not supported
SetForwarding	Sets up call forwarding.	Not supported
MonitorStart	Starts monitoring the telephone.	yes
MonitorStop	Stops monitoring the telephone.	yes
GetSwitchingFunctionDevices	Checks the configured identities.	yes
SnapShotDevice	Consistency check	yes
SystemRegister	Subscribes to notifications of configuration changes.	Not supported
SystemRegisterCancel	Unsubscribes to notifications of configuration changes.	Not supported
RequestSystemStatus	Checks the telephone status.	yes

3 Installation and configuration of uaCSTA Server for SIP Phones

3.1 Installation

Install uaCSTA Server for SIP Phones by running the uaCSTAServer_en-US.msi Setup program. Follow the instructions that will guide you through the installation process. Once the installation process has been completed, the wizard for the initial configuration of the driver will appear automatically. The individual wizard pages will be described by the following information. You can execute the administrator user interface from the Start Menu at any time.

The Admin Shell cuts into the following sections:
Phone connection
Client connection
Telephones
Advanced settings
Server status
Diagnostics
Information about

3.2 Phone connection

Telephones can contact uaCSTA Server for SIP Phones through two different network interfaces. Both interfaces support TCP and TLS connections. UDP is not supported because of problems with packet fragmentation in the large CSTA messages. For TLS connections, a certificate issued to the computer name must be provided, which trusts the telephones.
With the WAN interface, an IP address different from the DNS name can be stored in the port setting. Enter the public IP address (DNS-Name) port through which uaCSTA Server for SIP Phones will be accessible from the Internet. If the computer does not have a public IP interface itself, the entries will not be needed here.
You can only make changes to the connection settings when the Server Service has been shut down.

Login information for the telephones
When the telephone has been registered with uaCSTA Server for SIP Phones, it must use the access information configured here. The MAC address for the telephone will be expected as the user name. The password can be set from here. If you use the Provisioning feature provided in the administrator, the access information and connection parameters configured here will automatically be provided to the telephone.

3.3 Client connection

The ECSTA for SIP Phones TAPI drivers can contact uaCSTA Server for SIP Phones through two different network interfaces. Both interfaces support TCP and TLS connections. UDP is not supported because of problems with packet fragmentation in large CSTA messages. For TLS connections, a certificate issued to the computer name must be provided, which is trusted by the ECSTA for SIP Phones TAPI drivers.
With the WAN interface, an IP address different from the DNS name can be stored in the port setting. Enter the public IP address (DNS-Name) port through which uaCSTA Server for SIP Phones will be accessible from the Internet. If the computer does not have a public IP interface itself, the entries will not be needed here.
You can only make changes to the connection settings when the Server Service has been shut down.

Login information for the clients (ECSTA for SIP Phones)

You can configure the login information for the client connection from here. Specify this information during the configuration of the ECSTA for SIP Phones TAPI driver in the connection settings.

3.4   Access information

During the installation of uaCSTA Server for SIP Phones, you can configure the access information for ECSTA for SIP Phones and the telephones.

3.4.1   Login information for ECSTA for SIP Phones

You can configure the login information for the client connection from here. Specify this information during the configuration of the ECSTA for SIP Phones TAPI driver in the connection settings.

3.4.2   Login information for the telephones

If the telephone registers with uaCSTA Server for SIP Phones, the access information configured here must be used. The MAC address for the telephone will be expected as the user name. The password can be set from here. If you use the Provisioning feature provided in the administrator, the access information and connection parameters configured here will automatically be provided to the telephone.

3.5   Telephones

All telephones that are known to uaCSTA Server for SIP Phones are visible in the telephone list. The last identity used for login will be use listed in addition to the telephone type, the MAC address and the last known IP address. The last login and the status of the telephone provide information about if the telephone can be currently be controlled and monitored. Telephone will automatically appear in this list as soon as they have registered with the server for the first time. If you have activated the automatic activation of telephone under Advanced Options, the telephones will become available immediately for control and monitoring. If individual telephones are not provided for control and monitoring, uncheck the checkbox in the corresponding entries.

You can trigger Provisioning / Initial Configuration using the *Manage Telephones* button for additional telephones.

3.5.1   Manage Telephones - Provisioning / Initial Configuration

Provisioning is understood to be the automatic provision of the configuration for the telephones to be integrated. For this, the configuration parameters can be transferred directly to the telephones to be integrated from the administrator. In addition to the server name & IP address to be contacted, this includes the appropriate connection protocol (TCP, UDP, TLS), the user name and the password for logging in (regarding this, see phone connection).The configuration for the telephones will be made through the web interface. If special login information is required for access to the telephone, you can store this under *Options*. If you must provision special connection options, you can set them from *Options*.

Find Telephones
All locally accessible telephones will be found by a broadcast. If a telephone to be connected should not be found by this mechanism, you can manually add it by entering the IP address manually.

Providing Selected Telephones
Saves the provisioning information set under *Options* to the telephone. If necessary and activated under Options, the telephone will be re-started automatically by doing so.

Delete Provisioning
If you would like to delete the provisioning information on certain telephones, select them from the list and choose the *Delete Provisioning* menu item from the context menu. If necessary and activated under Options, the telephone will be re-started automatically by doing so.

If an automatic configuration for your telephones is not possible, you can find information about which settings must be made here:
Settings for Auerswald telephones
Settings for Mitel telephones
Settings for snom telephones
Settings for Yealink telephones
Settings for Polycom telephones
Settings forGigaset N870 IP PRO

3.5.1.1 Settings for Auerswald telephones

For the correct operation of the telephones at uaCSTA Server for SIP Phones, certain settings must be made in the telephone. Please consult the product documentation provided by the manufacturer together with the settings described here. The settings described here refer to a specific firmware version and may change during the course of further development. This information may therefore not be complete for the firmware that you use.

Settings for an Auerswald COMfortel - Firmware Version - March 2017

- Open the web page of the telephone.
- Add a new provider / PBX under Settings - Providers & PBX
- Select from the list of proposed profiles - **No Profile** - Import
- Provider - Settings for providers and PBX
  o Names: **uaCSTA**
  o Domain: DNS name or IPS address of the telephone interface in the uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
  o Type: **PBX**
- Provider - Registrar Settings
  o Registrar: **Activated**
  o Registrar address: DNS name or IP address of the telephone interface in uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
  o Registrar Port: Port of the telephone interface in uaCSTA Server for SIP Phones. **Example: 5060**
  o Time period for registration: **Example: 5**
- SIP - Settings for SIP
  o Server Type: **estos CSTA (ECSTA)**
- Add a new account under Settings - Accounts
- Settings for the selected account - Account
  o Account name: MAC address of the telephone without separators (you can get this via the Auerswald logo at the upper right area) **Example: 000952AABBCC**
  o Usage **Activated**
  o User name: MAC address of the telephone without separators **Example: 000952AABBCC**
  o Password: The password saved in uaCSTA Server for SIP Phones Administrator under *Telephone connection*
  o Authentication ID: MAC address of the telephone without separators **Example: 000952AABBCC**
  o Provider & PBX: **uaCSTA**
- **Settings - COMfortel - Extreme Control:**
  o uaCSTA: **Activated**

3.5.1.2 Settings for Mitel telephones

For the correct operation of the telephones at uaCSTA Server for SIP Phones, certain settings must be made in the telephone. Please consult the product documentation provided by the manufacturer together with the settings described here. The settings described here refer to a specific firmware version and may change

during the course of further development. This information may therefore not be complete for the firmware that you use.

Settings for a Mitel 68 - Firmware Version 4.1.0.2033 - February 2017
Mitel telephones must be provisioned via a configuration file together with the Web interface.

The following settings are to be made in the configuration file:
csta: 1
csta proxy: DNS name or the IP address of the telephone interface in uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
csta port: port of the telephone interface in uaCSTA Server for SIP Phones. **Example: 5060**
csta password: The password saved in uaCSTA Server for SIP Phones Administrator under *Telephone connection*

Example file:

```
csta: 1
csta proxy: uaCSTAServer.mydomain.com
csta port: 5060
csta password: passwort
```

Create a file with suitable content as startup.cfg in ASCII format. Save this file in a FTP, TFTP, HTTP(s) server in the root directory, e.g. Pumpkin TFTP Server

- Open the web page of the telephone.
- **Advanced Settings - Global SIP:**
  - Transport Protocol: **UDP and TCP**
- **Configuration Server:**
  - Enter the access data for the FTP, TFTP, HTTP(s) server
  - Save the settings
- **Operation - Reset:**
  - Restart phone - Restart the telephone

Check whether the telephone has adopted the settings correctly:

- Open the web page of the telephone.
- **Advanced Settings - Troubleshooting:**
  - Get server.cfg
  - The settings included in startup.cfg must be visible in the delivered file

3.5.1.3   *Settings for snom telephones*

For the correct operation of the telephones at uaCSTA Server for SIP Phones, certain settings must be made in the telephone. Please consult the product documentation provided by the manufacturer together with the settings described here. The settings described here refer to a specific firmware version and may change during the course of further development. This information may therefore not be complete for the firmware that you use.

Telephones from the snom manufacturer can also be conveniently provisioned via uaCSTA Server for SIP Phones Administrator via the interface. For this purpose, open the *Manage telephones* dialog under *Telephones*.

Settings for a snom821 - Firmware Version 8.7.5.35 - February 2017

- Open the web page of the telephone.

- Look for an identity that was not previously used, identity 12 in our example.
- Make the following settings:
- **Setup - Identity 12 - Login:**
  - Identity active: **An**
  - Displayed Name: **uaCSTA**
  - Username: MAC address of the telephone **Example: 000413AABBCC**
  - Password: The password saved in uaCSTA Server for SIP Phones Administrator under *Telephone connection*
  - Registrar: DNS-Name or IP address of the telephone interface in uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
  - Outbound Proxy: DNS-Name or IP address of the telephone interface in uaCSTA Server for SIP Phones followed by a port that may differ from the SIP standard and ;transport=TLS, **Example: uaCSTAServer.mydomain.com:5060;transport=TCP**
  - Hidden identity: An
- **Setup - Identity 12 - SIP:**
  - Validity range: **300**
- **Setup - Advanced - QoS/Security:**
  - Allow control via CSTA: **An**

Under settings, check whether *csta_legacy_control* is set to at least 3. If *csta_legacy_control* has a lower value, you must enter the following URL in the browser. This setting cannot be configured via the Web interface.
http://snom_ip/settings.htm?csta_legacy_control=4&settings=save

3.5.1.4   *Settings for Yealink telephones*

For the correct operation of the telephones at uaCSTA Server for SIP Phones, certain settings must be made in the telephone. Please consult the product documentation provided by the manufacturer together with the settings described here. The settings described here refer to a specific firmware version and may change during the course of further development. This information may therefore not be complete for the firmware that you use.

Settings for a Yealink T46G - Firmware Version 28.81.0.15 - February 2017

- Open the web page of the telephone.
- Look for an account that was not previously used, account 16 in our example.
- Make the following settings:
- **Account - Account 16 - Register:**
  - Account active: **Activated**
  - Display-Name: **uaCSTA**
  - Registered name: MAC address of the telephone **Example: 001565AABBCC**
  - User Name: MAC address of the telephone **Example: 001565AABBCC**
  - Password: The password saved in uaCSTA Server for SIP Phones Administrator under *Telephone connection*
  - Server Host: DNS-Name or IP address of the telephone interface in uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
  - PriChat: Port of the telephone interface in the uaCSTA Server for SIP Phones. **Example: 5060**
  - Transport: Protocol of the telephone interface in uaCSTA Server for SIP Phones. **Example: TCP**
  - Server running from: **300**
- **Performances - Remote Control:**
  - CSTA Control: **Activated**

3.5.1.5   Settings for Polycom telephones

For the correct operation of the telephones at uaCSTA Server for SIP Phones, certain settings must be made in the telephone. Please consult the product documentation provided by the manufacturer together with the settings described here. The settings described here refer to a specific firmware version and may change during the course of further development. This information may therefore not be complete for the firmware that you use.

Settings for a Polycom VVX 411 oder VVX 350 - Firmware Version 6.1.0.6189 - October 2019

- Open the web page of the telephone.
- Search for a line which has not been used thus far, in our example Line 2
- Make the following settings:
    o **Select identification**
    o Display name: **uaCSTA**
    o Address: MAC address of the telephone **Example: 64167F011214**
    o **Select authentication**
    o User ID: MAC address of the telephone **Example: 64167F011214**
    o Password: the password stored in the uaCSTA Server for SIP Phones administrator under *Telephone Connection*
    o **Select server 1**
    o Special-Interop: **CSTA**
    o Address: DNS name or IP address of the telephone interface in uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
    o Port: port of the telephone interface in uaCSTA Server for SIP Phones. **Example: 5060**
    o Network transport: Protocol of the telephone interface in uaCSTA Server for SIP Phones. **Example: TCP**
    o Sequence (s): **300**
    o Register: **Yes**

3.5.1.6   Settings forGigaset N870 IP PRO

For the correct operation of the DECT telephones connected to the Gigaset N870 IP PRO on uaCSTA Server for SIP Phones, certain settings must be made on the Gigaset N870 IP PRO. In addition to the settings described here, please also consult the product documentation provided by the manufacturer. The settings described here relate to a specific firmware version and may change in the course of the further development of the telephones. This information may therefore not be complete for the firmware you are using.

Settings for Gigaset N870 IP PRO - Firmware Version 2.39.0

- o Open the website of the Gigaset N870 IP PRO.
    o Select Provider or PBX Profiles and the profile for editing via which you have configured your VOIP provider.
    o Go to the CSTA section and make the following settings:
        ▪ Active: select **yes**
        ▪ Extern: select **yes**
        ▪ Address: DNS name or IP address of the telephone interface in uaCSTA Server for SIP Phones. **Example: uaCSTAServer.mydomain.com**
        ▪ Port: Port of the telephone interface in uaCSTA Server for SIP Phones. **Example: 5060**
        ▪ Elapse time (s): **Example: 300**
        ▪ DNS query: select **A**
        ▪ Network transport: Protocol of the telephone interface in uaCSTA Server for SIP Phones. **TCP**
    o Select handsets.

- o Create a new handset or select an already created handset for editing.
- o Go to the CSTA section and make the following settings:
  - **Select authentication**
  - Username: A character string according to a MAC address convention **(example: 589EC6010D58)**. The first 6 characters must be 589EC6 or 7C2F80.
  - Login name: The same string as the user name.
  - Password: The password stored in the uaCSTA Server for SIP Phones administrator under *Telephone connection*

3.6   Advanced settings

Automatically put phones into operation
If you have activated this option, uaCSTA Server for SIP Phones will automatically place unknown telephones into operation as soon as they have registered with the server for the first time. The telephone will be immediately available for control and monitoring thereafter.

SIP Expiration Time (in seconds)
Defines the expiration time suggested by the uaCSTA proxy for SIP registration and SIP dialogs. Modify these settings only in case of need.

Keep Connection Alive Time (in seconds)
Defines the intervals in which CSTA Keep Alive messages will be exchanged between the server and the telephone. Modify these settings only in case of need.

3.7   Server status

In this dialog you can see the status of the server service and you can start and stop this service here. uaCSTA Server for SIP Phones will run as a Windows® system service. The service will be registered during installation and started after it has been configured. The Diagnostics window will display the log file from the last time the service was started.

3.8   Diagnostics

From this dialog, you can configure the logging options for the diagnosis of problems.

Log Level
Here you can select how much information is written to the log files.

Maximum size of a log file
Several log files are written. Each log file is created periodically when the set value here is exceeded.

Delete log files daily
If this option is active, the log files will be deleted daily.

Log file directory
In this directory the log files are stored. Note that the service requires write permission to this directory.

Log Filter - Telephones (MAC addresses)
In order to be able to analyze problems with individual telephone better, logging can be limited to specific telephones. Enter the MAC addresses for the telephones to be logged from here. Use a comma as the separator (AABBCCDDEEFF, 001122334455).

4   Info about estos uaCSTA Server for SIP Phones

estos uaCSTA Server for SIP Phones is a product of estos GmbH.

Product updates can be found under http://www.estos.de.

Frequently asked questions and answers, as well as support, can be found under https://www.estos.de/service.

All trade names referenced are the service marks, trademarks or registered trademarks of their respective companies.

4.1.1   Communications between telephones, server and clients

The following diagram will show the course of communications between the telephone, the server and a connected client based on ECSTA for SIP Phones in a heavily simplified manner.
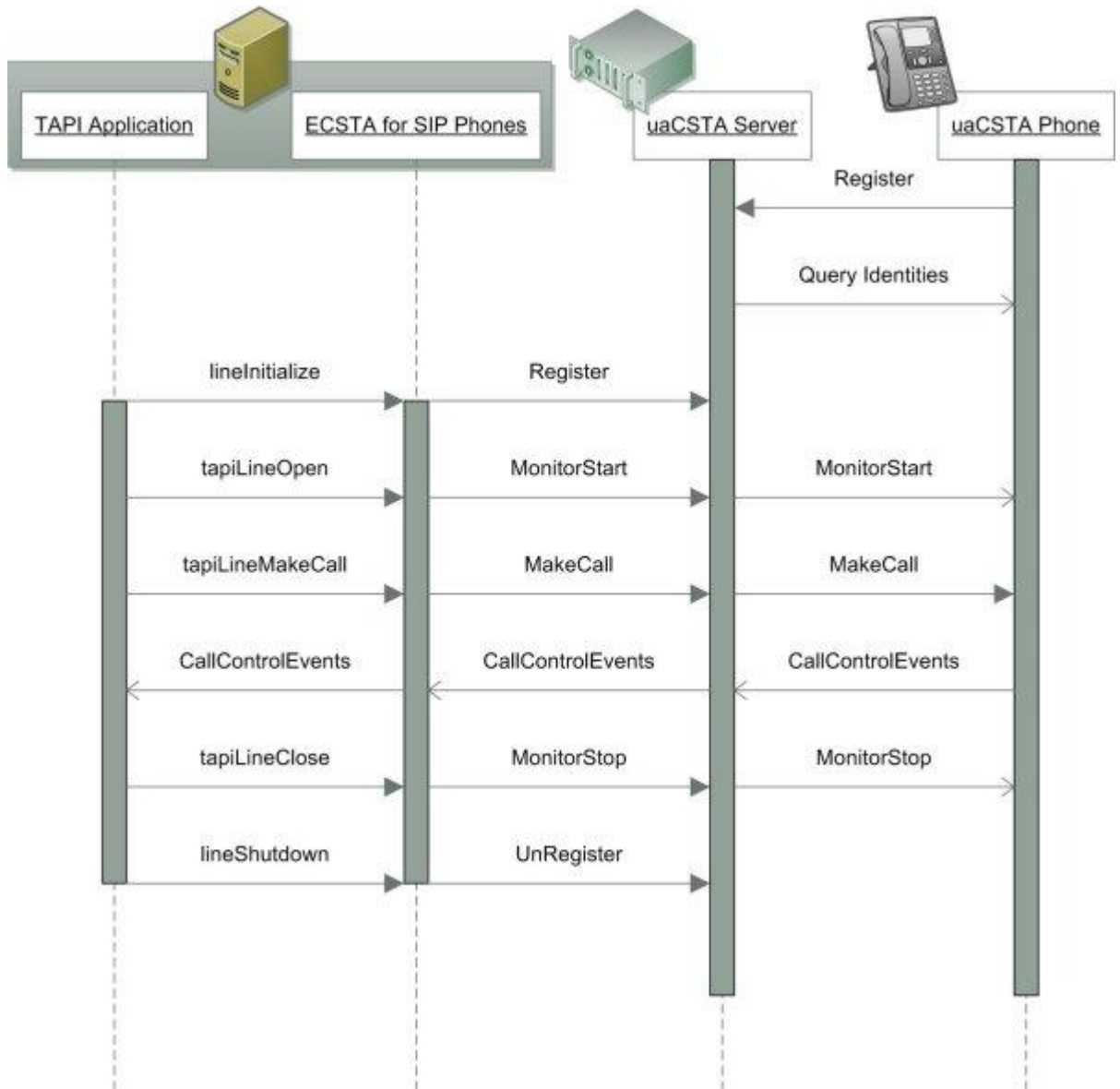
Abstract description of the message flows:

After the telephone has been authenticated on the server through the SIP register, the server will create an SIP dialog for CSTA communications with the telephone. The server will register for messages about configuration changes through this SIP dialog and determine the currently configured identities. The SIP dialog will stay open continuously.

As soon as a TAPI application wants to monitor and control a telephone registered with the server, the client will be registered with uaCSTA Server for SIP Phones. The line will be open at the same time on the server. The server will asynchronously open the CSTA monitor point in the telephone and report as soon as the telephone is being monitored through a BackInService event. The client will then query all additional information relevant for it (call forwarding configuration, DND status, message waiting, etc.)
All line control functions will be called synchronously. A MakeCall message will be sent to the associated telephone over uaCSTA Server for SIP Phones and returned after successful execution. CallControl events will be sent in the opposite direction from the telephone through the server to all associated clients. Closing the line in the client will delete the monitoring point in the telephone, insofar as additional interest is not reported.

4.2 Technical documentation

The uaCSTA server can be used for CSTA communications with telephones by third-party applications. This section will describe the technical foundations of communications as well as the uaCSTA features supported by the server.

- o Communication between Telephones, Server and Clients
- o uaCSTA Features

4.2.1 uaCSTA Features

This list describes all CSTA methods and events supported by estos uaCSTA Server for SIP Phones. After the server has been added as a proxy for CSTA messages, the telephone must actually support the feature. The features supported by the telephones will be described under Prerequisites.

CSTA Area	CSTA Methods / Events	Service	Event
Capability Exchange Services / Events:			
	GetLogicalDeviceInformation	X	
	GetPhysicalDeviceInformation	X	
	GetSwitchingFunctionDevices	X	
	SwitchingFunctionDevices		X
System Services / Events:			
	SystemRegister	X	
	SystemRegisterAbort		X
	SystemRegisterCancel	X	
	RequestSystemStatus	X	
	SwFunctionDevicesChanged		X
Monitoring Services:			
	MonitorStart	X	
	MonitorStop	X	X
SnapShot Services:			

	SnapshotDevice	X	
	SnapshotCall	X	
Call Control Services / Events:			
	AlternateCall	X	
	AnswerCall	X	
	ClearConnection	X	
	ConferenceCall	X	
	ConsultationCall	X	
	DeflectCall	X	
	DialDigits	X	
	DirectedPickupCall	X	
	HoldCall	X	
	MakeCall	X	
	ReconnectCall	X	
	RetrieveCall	X	
	SingleStepConferenceCall	X	
	SingleStepTransferCall	X	
	TransferCall	X	
	CallCleared		X
	Conferenced		X
	ConnectionCleared		X
	Delivered		X
	DigitsDialed		X

	Diverted		X
	Established		X
	Failed		X
	Held		X
	NetworkCapabilitiesChanged		X
	NetworkReached		X
	Offered		X
	Originated		X
	Queued		X
	Retrieved		X
	ServiceInitiated		X
	Transferred		X
			X
Call Associated Features Services:			
	GenerateDigits	X	
	DigitsGenerated		X
Physical Device Feature Services:			
	GetMessageWaitingIndicator	X	
	SetMessageWaitingIndicator	X	
	MessageWaiting		X
Logical Device Features Services / Events:			
	GetDoNotDisturb	X	
	GetForwarding	X	

	SetDoNotDisturb	X	
	SetForwarding	X	
	DoNotDisturb		X
	Forwarding		X