OpenScape 4000
OpenScape SBC

How to Configure htp Business FleX SIP-Trunk smart of htp GmbH

Table of Contents

Table of History

Date	Version	Changes
17.11.2022	1.0	Initial version

Trunk Configuration Data provided by htp

The SIP trunk configuration data (IP addresses etc.) needed to setup the SIP trunk can be found in the htp interconnection document provided by htp.

(https://www.htp.net)

OpenScape 4000 Configuration

In this section the typical AMO (Administration and Maintenance Order) commands to create a native SIP trunk between htp SSP and OS4000 vHG board will be described as an example.

See also OpenScape 4000 Service Documentation – IP Solutions

The AMO commands are executed through ComWin application that interfaces to OS4000's database.

Add function block for vHG board:

```
ADD-BFDAT:FCTBLK=6,FUNCTION=HG3550,BRDBCHL=BCHL120,ATTR=SOCO;

CHANGE-BFDAT:CONFIG=CONT,FCTBLK=6,FUNCTION=HG3550,LINECNT=4,UNITS=3;

CHANGE-BFDAT:CONFIG=OK,FCTBLK=6,ANSW=YES;
```

Add vHG board in the SWU:

```
ADD-BCSU:MTYPE=IPGW,LTG=1,LTU=99,SLOT=1,PARTNO="Q2330-
X",FCTID=1,LWVAR="0",FCTBLK=6,BCHL3550=30,ALARMNO=0,IPMODE=IPV4,DHCPV4=NO,DHCPV6=NO;
```

Configure vHG board data (device specific parameters e.g., board IP, default gateway, assign number of SIP channels etc.):

```
ADD-CGWB:LTU=99,SLOT=1,SMODE=NORMAL,IPADR=10.8.242.115,NETMASK=255.255.255.0;

CHANGE-
CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=GLOBIF,PATTERN=213,VLAN=NO,VLANID=0,DEFRT=10.8.242.1,TRPRSIP
=120,
TRPRSIPQ=0,TRPRH323=0,TPRH323A=0,TLSP=4061,DNSIPADR=10.8.251.103,DNSIPAD2=0.0.0.0,USEWANIF=NO,
WPUBIP=0.0.0.0,SIPTCPP=5060,SIPTLSP=5061;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=SERVIF,LOGINTRM="TRM",PASSW="HICOM";

CHANGE-
CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,UDPPRTLO=29100,UDPPRTHI=30099,TOSPL=184,TOSSIGNL=104,T38
FAX=NO,  RFCFMOIP=NO,RFCDTMF=YES,REDRFCTN=NO,PRIO=PRIO1,CODEC=G711A,VAD=NO,RTP=30;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO2,CODEC=G729A,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO3,CODEC=NONE,VAD=NO,RTP=30;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO4,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO5,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO6,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO7,CODEC=G729AB,VAD=YES,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO8,CODEC=G722,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=ASC,PRIO=PRIO9,CODEC=OPUS,VAD=NO,RTP=20;

CHANGE-
CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=MGNTDATA,MGNTIP=10.8.242.100,MGNTPN=8000,BUSIP=10.8.242.100,
BUSPN=443,UIMODE=CLASSIC;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=DMCDATA,DMCCONN=0,SMP=YES,SMP4OSV=NO;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-DEVEL",ROLE=ENGR;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-SU",ROLE=SU;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-ADMIN",ROLE=ADMIN;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-READER",ROLE=READONLY;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=GWDATA,GWID1="PRIMARYRASMANAGERID";
```

```
CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=SIPTRERH,GWAUTREQ=NO;

CHANGE-
CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=SIPTRSSA,SIPREG=NO,REGIP1=0.0.0.0,PORTTCP1=5060,PORTTLS1=506
1, REGTIME=300,REGIP2=0.0.0.0,PORTTCP2=5060,PORTTLS2=5061;

CHANGE-
                              CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=DLSDATA,DLSIPADR=10.6.25.5,DLS
                              PORT=18443,DLSACPAS=YES;

CHANGE-
CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=JB,AVGDLYV=40,MAXDLYV=120,MINDLYV=20,PACKLOSS=4,AVGDLYD=60,
MAXDLYD=200,JBMODE=2;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=IPCONF,IPMODE=IPV4,DHCPV4=NO,DHCPV6=NO;

CHANGE-
CGWB:MTYPE=CGW,LTU=99,SLOT=1,TYPE=MANLANIF,MIPADR=0.0.0.0,MNETMASK=0.0.0.0,MVLAN=NO,MVLANID=0,
MDEFRT=0.0.0.0;
```

Add Class Of Parameter (used to specify the line parameters for signaling control of the device handler)**:**

```
ADD-COP:COPNO=1,PAR=ANS&L3AR&IMEX,TRK=TA,TOLL=TA;

CHANGE-COP:COPNO=1,COPTYPE=COPADD,DEV=INDEP,INFO="IP TR";
```

Add Class Of Trunk (used to specify the switching technology parameters)**:**

```
ADD-
COT:COTNO=1,PAR=RCL&IIDL&IVAC&INAU&ANS&CHRT&AEOD&CEBC&COTN&IEVT&IDIS&BSHT&BLOC&PROV&LWNC&INDG&
NLCR&TSCS& ICZL&ABNA&ABPD&WAAN&DFNN&IONS&NLRD&NOFT&AMFC&NPIS&NTON;
```

Add Class Of Service (used to specify the authorizations and features assigned to stations and trunks)**:**

```
ADD-COSSU:NEWCOS=1,INFO="";

CHANGE-COSSU:TYPE=COS,COS=1,AVCE=TA&TNOTCR&CDRINT&COSXCD&MB&DATA&CFNR&VCE;

CHANGE-COSSU:TYPE=COS,COS=1,AVCE=RSVLN&DICT&SPKR&FWDNWK&TTT&MSN&CFB&MULTRA;

CHANGE-COSSU:TYPE=COS,COS=1,AVCE=FWDEXT&CCBS&CW&GRPCAL;

CHANGE-COSSU:TYPE=COS,COS=1,AFAX=TA&TNOTCR;

CHANGE-COSSU:TYPE=COS,COS=1,ADTE=TA&TNOTCR&CDRINT&BASIC&MSN&MULTRA;
```

Add Bundle (used to specify trunk group number identifier)**:**

```
ADD-BUEND:TGRP=2,NAME="htp
SSP",NO=30,TRACENO=0,ACDTHRH=*,PRIONO=2,TDDRFLAG=OFF,GDTRRULE=0,ACDPMGRP=0, CHARCON=NEUTRAL;
```

Add digital trunk circuits in the SWU (used for the external gateway for IP trunking configuration)**:**

```
ADD-TDCSU:OPT=NEW,PEN=1-99-001-0,COTNO=1,COPNO=1,DPLN=0,ITR=0,COS=1,LCOSV=1,LCOSD=1,CCT="htp
SSP",
DESTNO=0,PROTVAR=ECMAV2,SEGMENT=8,DEDSVC=NONE,TRTBL=GDTR,SIDANI=N,ATNTYP=CO,CBMATTR=NONE,TCHAR
G=N,SUPPRESS=0,ISDNIP=00,ISDNNP=0,TRACOUNT=30,SATCOUNT=MANY,ALARMNO=0,FIDX=1,CARRIER=1,ZONE=EM
PTY,COTX=1,FWDX=5,CHIMAP=N,
UUSCCX=16,UUSCCY=8,FNIDX=0,NWMUXTIM=10,SRCGRP=5,CLASSMRK=EC&G711&G729AOPT,TCCID="",TGRP=2,SRCH
MODE=DSC,INS=Y,SECLEVEL=TRADITIO,HMUSIC=0,CALLTIM=60,WARNTIM=60,DEV=HG3550CO,BCHAN=1&&15,BCNEG
=N,BCGR=1,LWPP=0,LWLT=0, LWPS=0,LWR1=0,LWR2=0,DMCALLWD=N,GWPROT=NONE;
```

Add digit analysis (point access code to a suitable route)**:**

```
ADD-WABE:CD=17,DAR=NETRTE;
```

Add SIP trunk destination (used for allocating SIP trunk access codes)**:**

```
ADD-RICHT:MODE=LRTENEW,LRTE=2,LSVC=ALL,NAME="htp SSP",TGRP=2,DNNO=1-1-
122,ROUTOPT=NO,DTMFCNV=FIX,DTMFTEXT="",
DTMFPULS=PP80,ROUTATT=NO,EMCYRTT=NO,INFO="",PDNNO=0,CHARCON=NEUTRAL,CONFTONE=NO,RERINGRP=NO,NO
PRCFWD=NO,NITO=NO,CLNAMEDL=NO,FWDSWTCH=NO,LINFEMER=NO,NOINTRTE=NO;
```

For MWI:

```
ADD-RICHT:MODE=PM,IDX=1,SAN=49511920946309,NAME="XPR",STYPE=XPRESION;
```

Add LCR outdial rule (used to specify outdial rule number identifier)**:**

```
ADD-LODR: ODR=2,CMD=NPI,NPI=ISDN,TON=INTERNAT;
```

```
ADD-LODR:ODR=2,CMD=ECHO,FIELD=2;
```

```
ADD-LODR:ODR=2,CMD=END;
```

```
ADD-LODR:ODR=2,INFO="htp SSP";
```

Add Administration of LCR (used to specify the accumulation of the trunk groups)**:**

```
ADD-
LDAT:LROUTE=2,LSVC=ALL,LVAL=1,TGRP=2,ODR=2,LAUTH=1,CARRIER=1,ZONE=EMPTY,LATTR=WCHREG,VCCYC=4;
```

Add digits to dial plan (refers to the digit analysis results of a dialed digit sequence or dialed code):

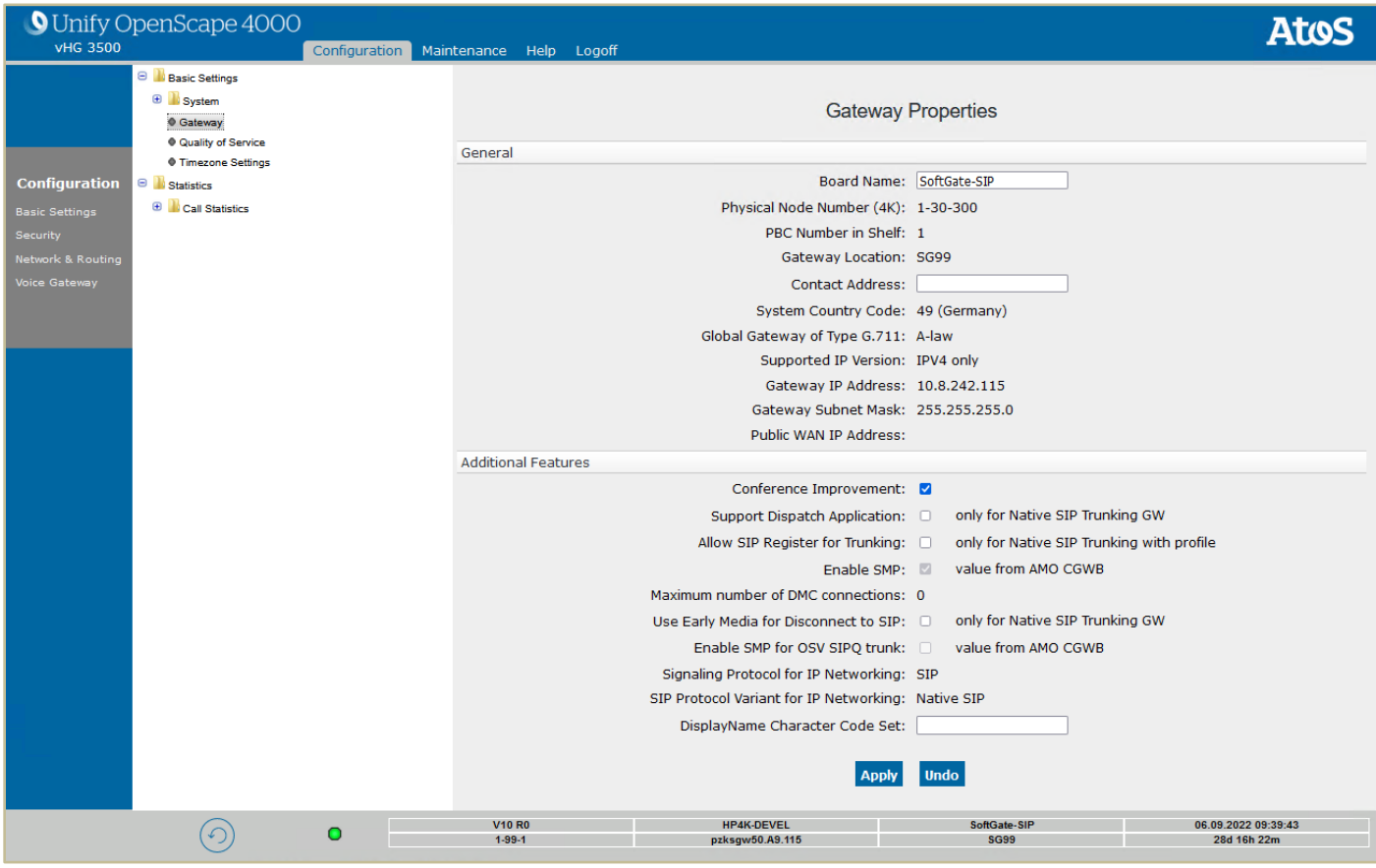
```
ADD-WABE:CD=9,DAR=CO,CHECK=N;
```

Add Administration LCR – Dialplan (used to specify the LCR digit patterns - LDP):

```
ADD-LDPLN: LCRCONF=LCRPATT,DIPLNUM=0,LDP="9"-"Z",DPLN=0&1&2&3&4&5&6&7&8&9&10&11&12&13&14&15,
LROUTE=2,LAUTH=1,PINDP=N;
```

WBM Configuration for htp Native SIP Trunk - Gateway Properties

Navigate to HG WBM >> Configuration >> Basic Settings >> Gateway.

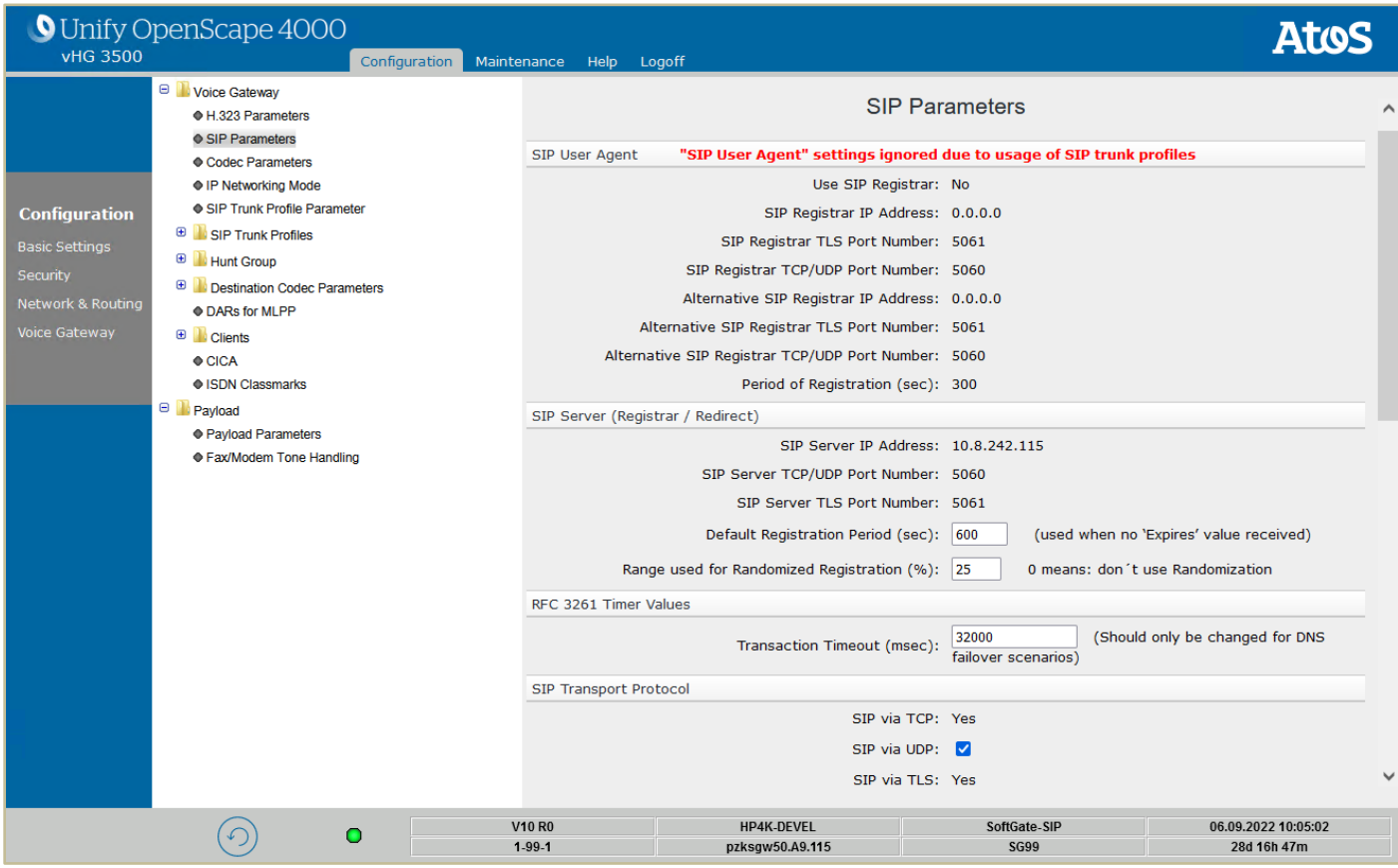


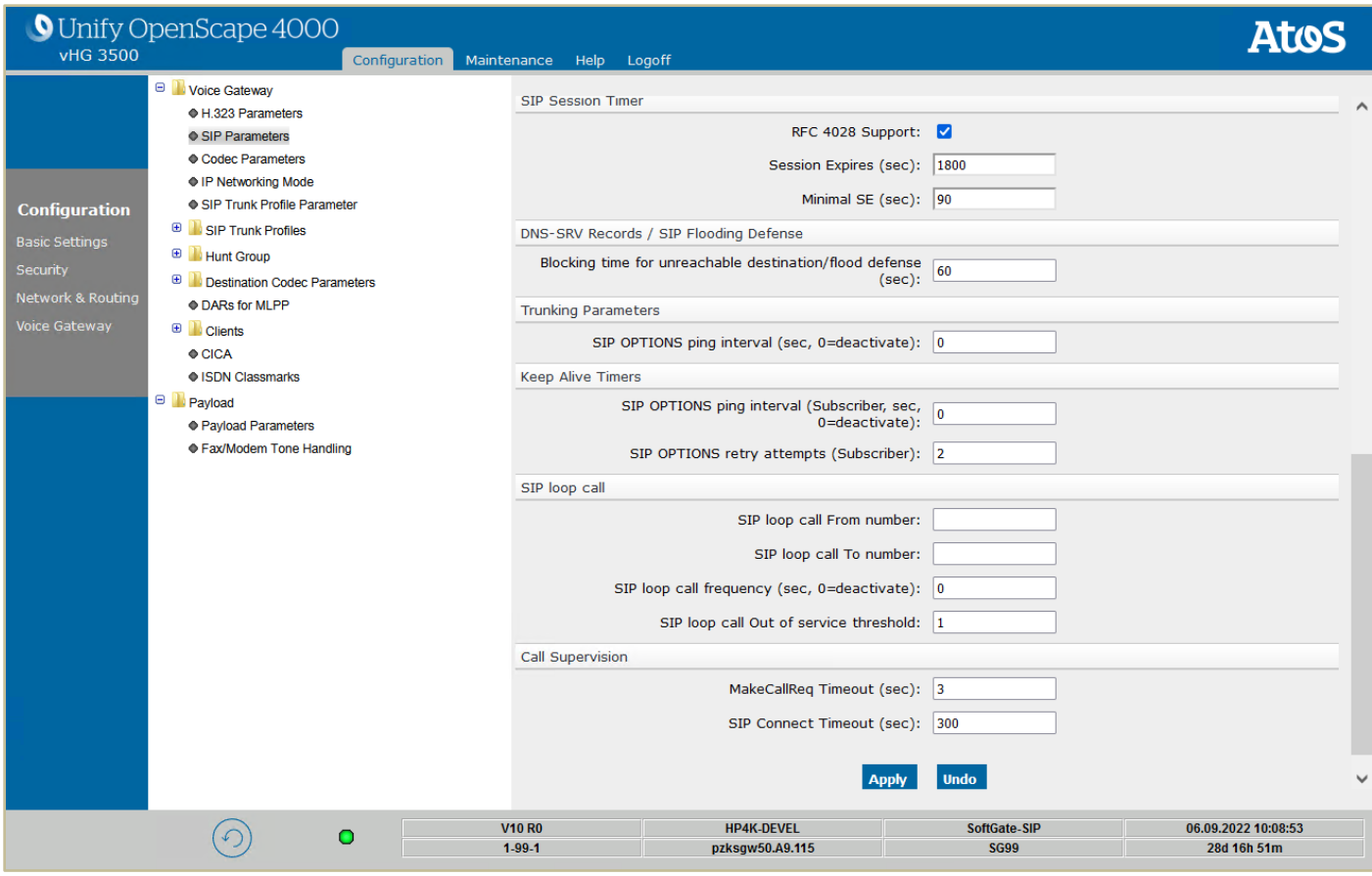
Make sure that:

- **Signaling Protocol for IP Networking**: SIP
- **SIP Protocol Variant for IP Networking:** Native SIP

WBM Configuration for htp Native SIP Trunk - SIP Parameters

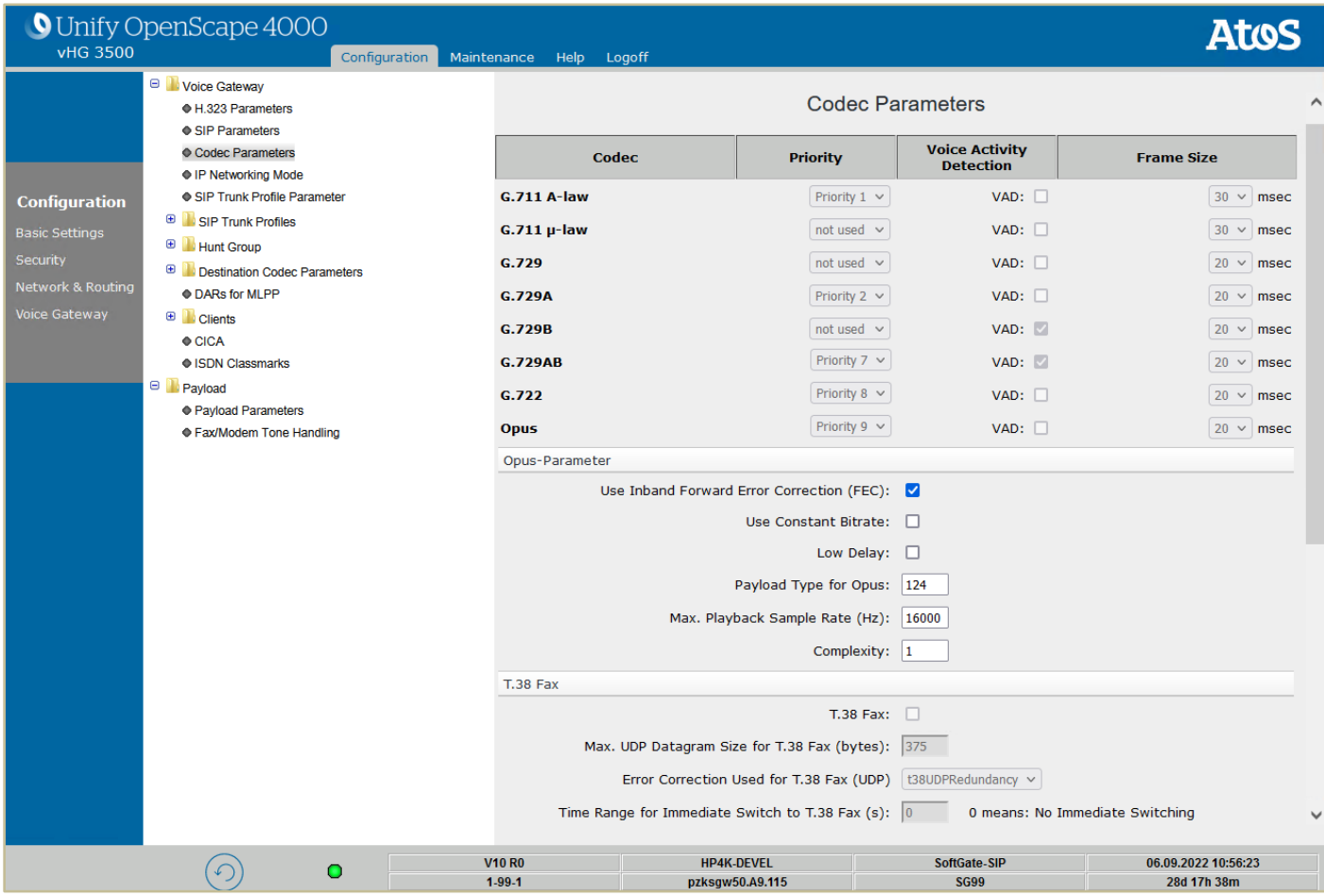
The vHG **"SIP Parameters"** used for the certification activities are shown under **HG WBM >> Configuration >> Voice Gateway >> SIP Parameters**.

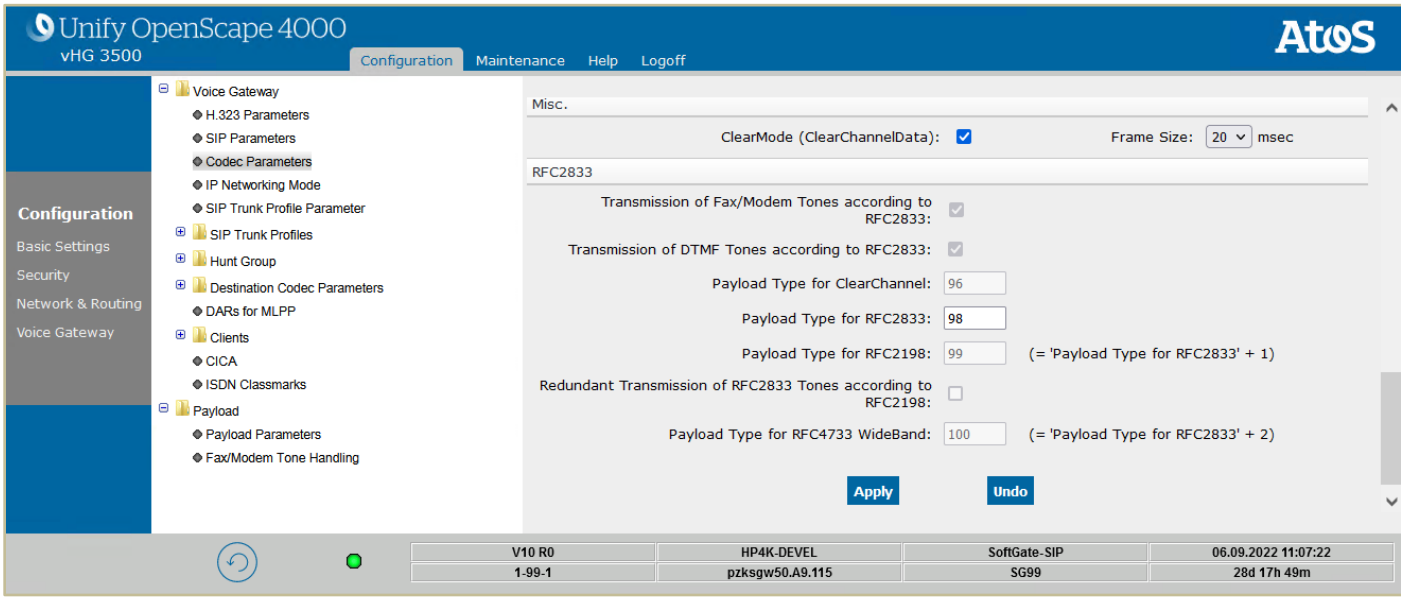



Unify OpenScape 4000
vHG 3500
Configuration
Maintenance
Help
Logoff
Atos
Configuration
Basic Settings
Security
Network & Routing
Voice Gateway
Voice Gateway
H.323 Parameters
SIP Parameters
Codec Parameters
IP Networking Mode
SIP Trunk Profile Parameter
SIP Trunk Profiles
Hunt Group
Destination Codec Parameters
DARs for MLPP
Clients
CICA
ISDN Classmarks
Payload
Payload Parameters
Fax/Modem Tone Handling
SIP Session Timer
RFC 4028 Support:
Session Expires (sec): 1800
Minimal SE (sec): 90
DNS-SRV Records / SIP Flooding Defense
Blocking time for unreachable destination/flood defense (sec): 60
Trunking Parameters
SIP OPTIONS ping interval (sec, 0=deactivate): 0
Keep Alive Timers
SIP OPTIONS ping interval (Subscriber, sec, 0=deactivate): 0
SIP OPTIONS retry attempts (Subscriber): 2
SIP loop call
SIP loop call From number:
SIP loop call To number:
SIP loop call frequency (sec, 0=deactivate): 0
SIP loop call Out of service threshold: 1
Call Supervision
MakeCallReq Timeout (sec): 3
SIP Connect Timeout (sec): 300
Apply
Undo
V10 R0
HP4K-DEVEL
SoftGate-SIP
06.09.2022 10:08:53
1-99-1
pzksgw50.A9.115
SG99
28d 16h 51m

WBM Configuration for htp Native SIP Trunk - Codec Parameters

Go to **HG WBM >> Configuration >> Voice Gateway >> Codec Parameters** to view the vHG **"Codec Parameters"** utilized for the current testing environment. As an example:

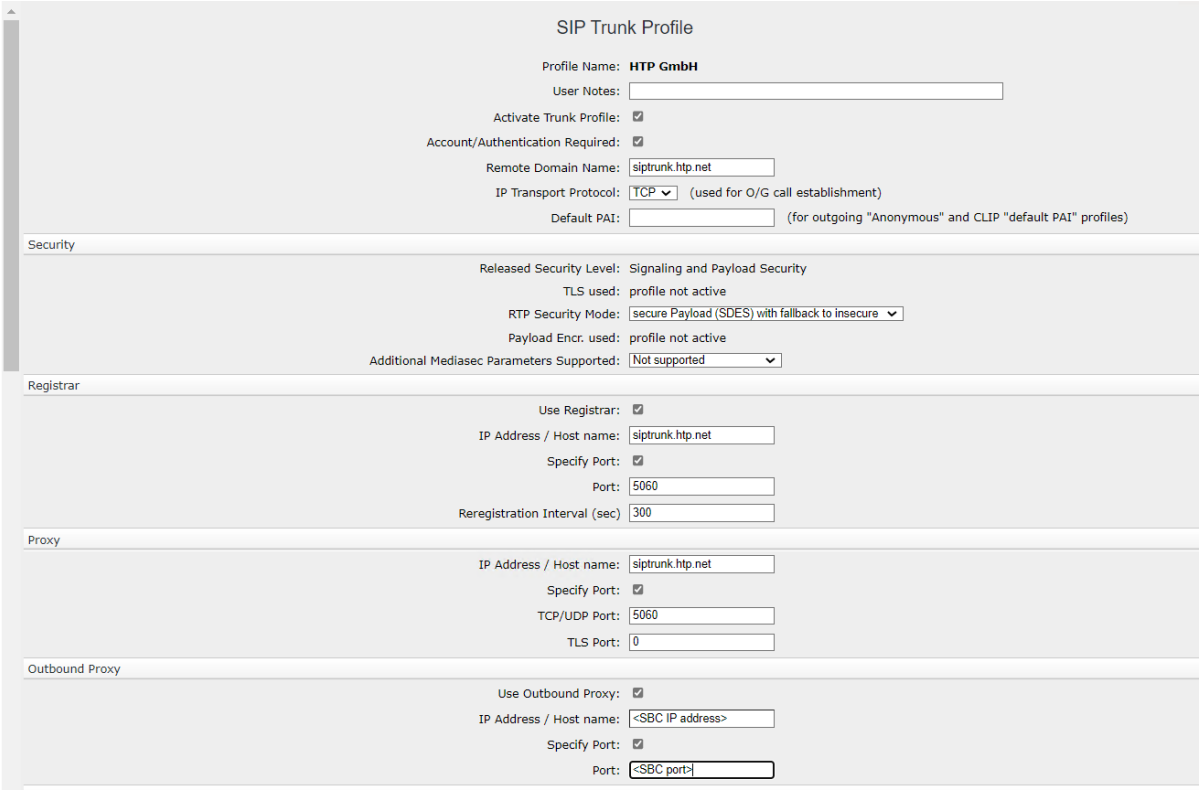




Note 1: The greyed out options can be changed via AMO CGWB.

Note 2: In real-life production environments htp requires **"Frame Size"** value to be set to "20".

SIP trunk Profile:



OpenScape SBC Configuration

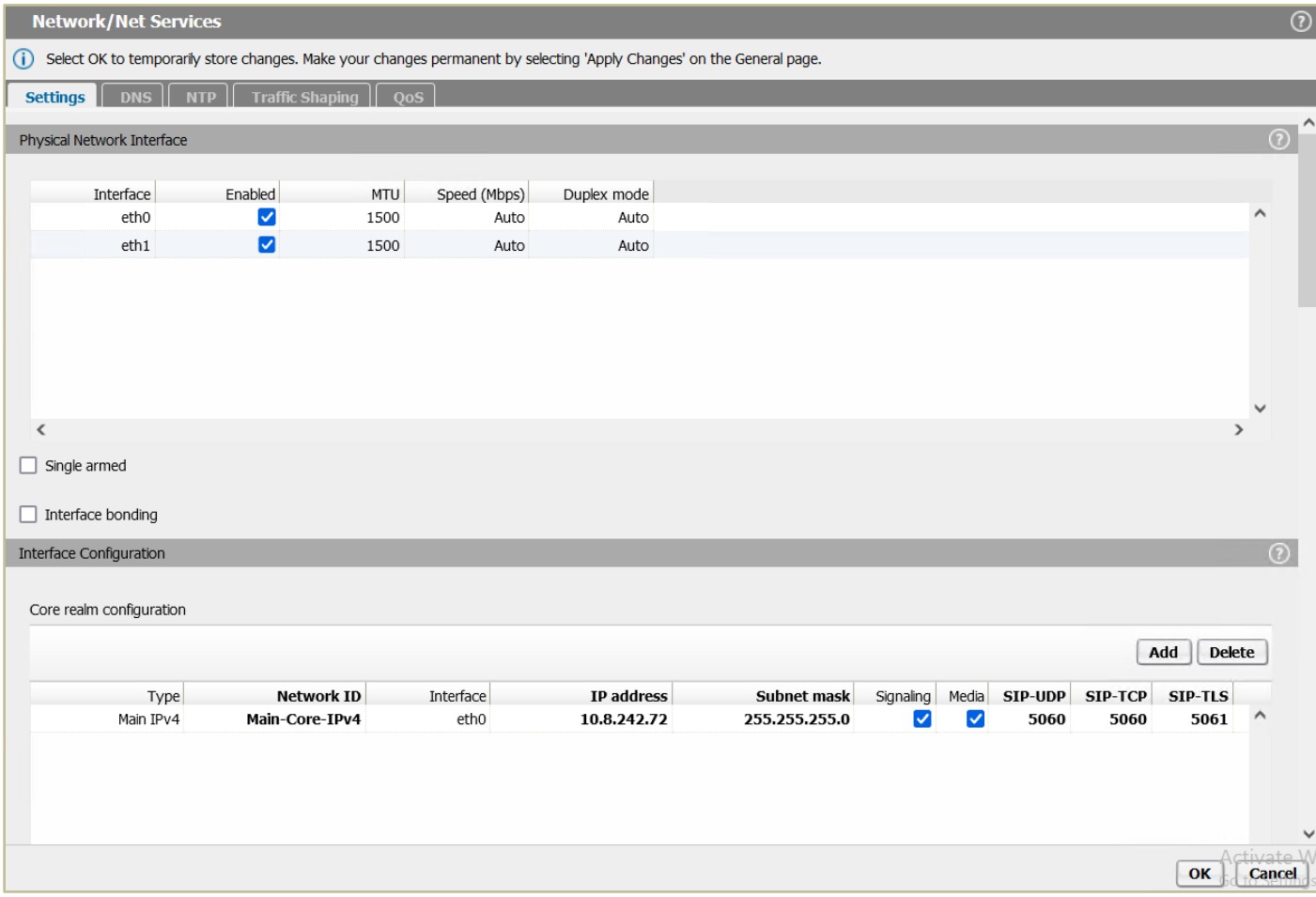
This chapter outlines the configuration of OpenScape SBC for interworking with htp Enterprise Voice network.

The OpenScape SBC will be configured with the connection to OS4000 and SSP (htp) endpoints. Routine or non-project specific OS SBC configuration will be omitted.

Core Realm Interface

Use the TCP ("Proxy") port number configured as outbound proxy in the OS4K WBM for the connection of OS SBC's eth0 (core) interface to OS4000.

Go to **OS SBC Management Portal >> Network/Net Services.**



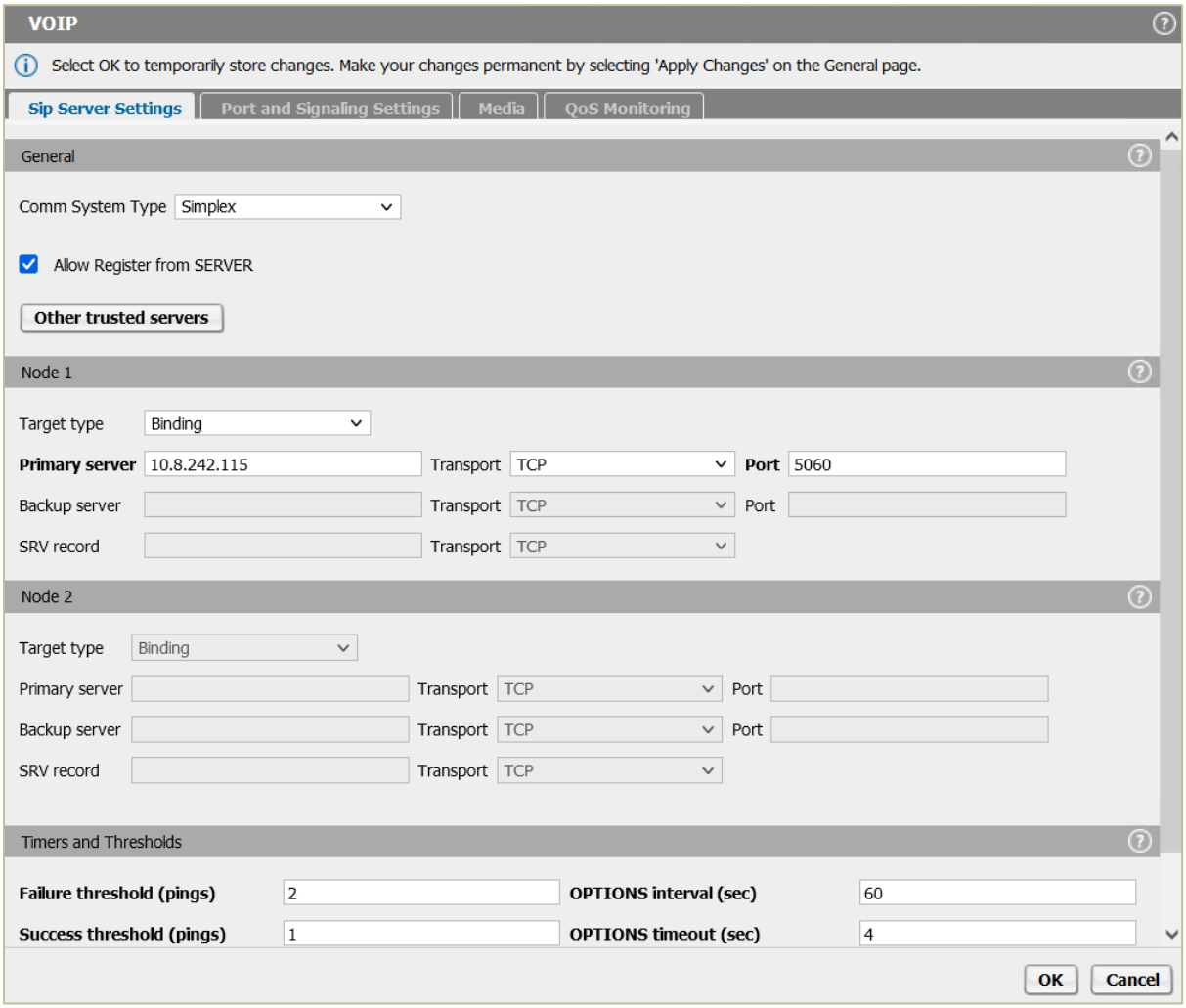
On **"Settings"** tab and in **"Core realm configuration"** area, make sure that for **"eth0"** interface, **"SIP-TCP"** has the value "5060".

Click on **[OK]**.

Click on **[Apply Changes]** on OS SBC main page.

SIP Server

The SIP connectivity to OS4000 is configured in **OS SBC Management Portal >> VOIP** window.



On **"Sip Server Settings"** tab, enter the following:

- **Comm System Type**: Simplex
- **Allow Register from SERVER:** Enabled
- **Target Type**: Binding
- **Primary Server:** (OS4000 HG card IP address of the trunk)
- **Transport:** TCP
- **Port:** 5060 (listening port)

Click on **[OK]**.

Click on **[Apply Changes]** on OS SBC main page.

Certificates

In case TLS interconnection is required between OpenScape SBC and htp systems, 1 file in pem format is required in OS SBC:

CA certificate (e.g., serverCA.pem)

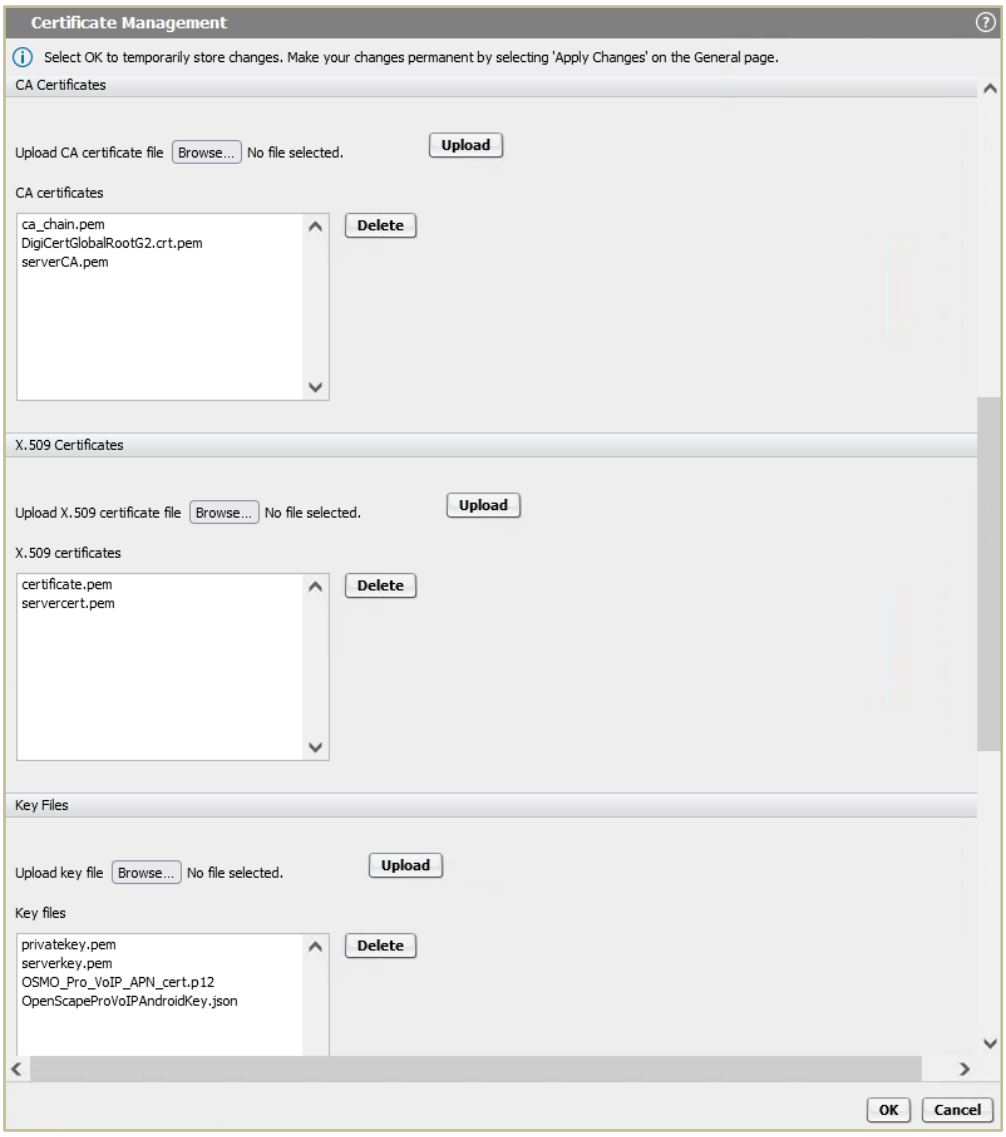
Server certificate for OS SBC (e.g., servercert.pem)

OS SBC server certificate private key used for the CSR to CA (e.g., serverkey.pem)

In current certification activities the default certificates and private key were used. Custom certificate and private key files may be used after uploading these files to SBC. For the TLS negotiation the SBC operates as a client and htp as a server, therefore the certificate provided by htp should be included in OS SBC.

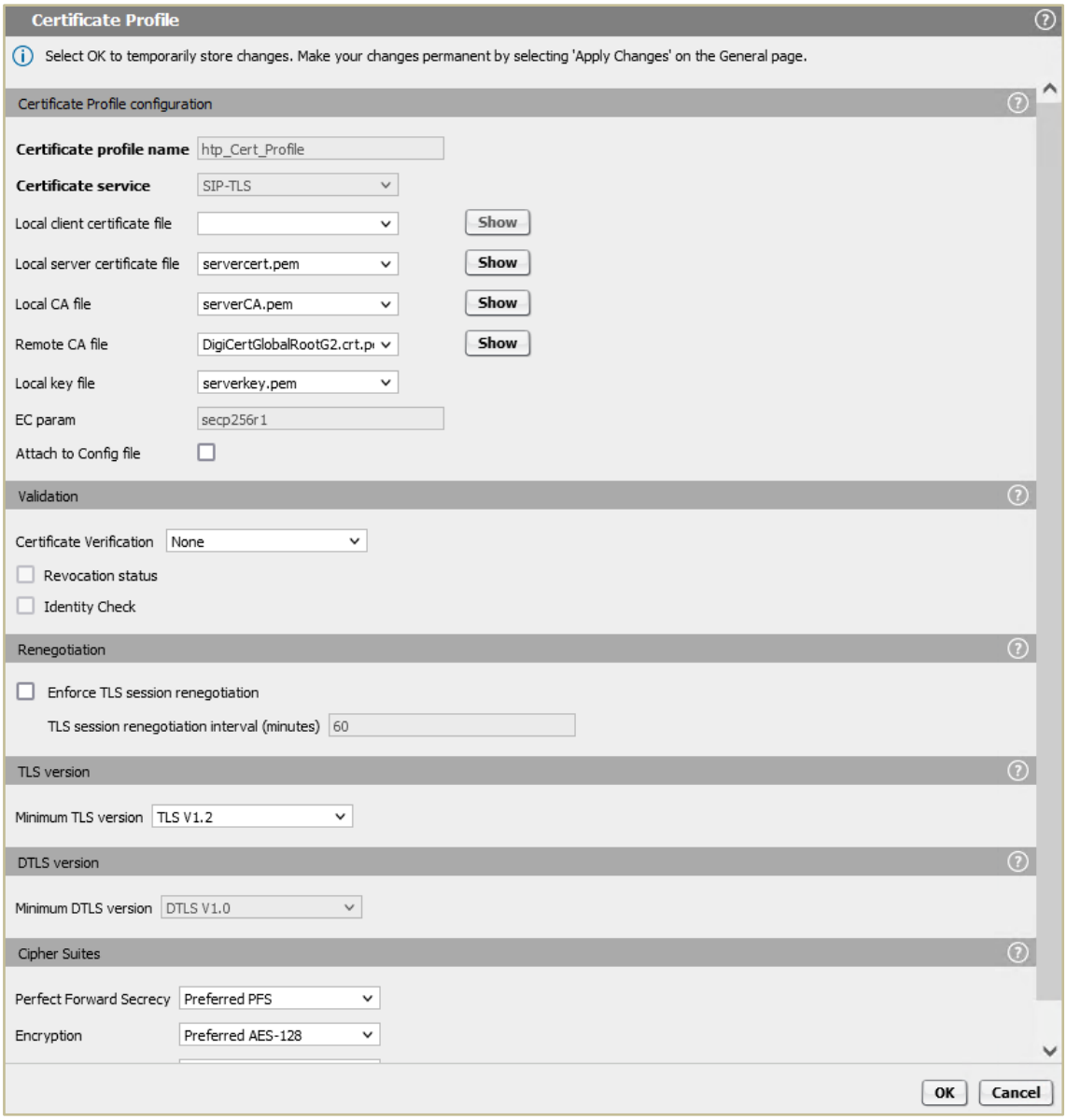
Navigate to **OS SBC Management Portal >> Security >> General** and click on **"Certificate Management"** button.

In case custom OS SBC certificates are required, the corresponding files may be uploaded from **"CA Certificates"**, **"X.509 Certificates"** and **"Key Files"** areas correspondingly, as shown in figure below:



The certificate provided by htp, "DigiCertGlobalRootG2.crt.pem", is also uploaded.

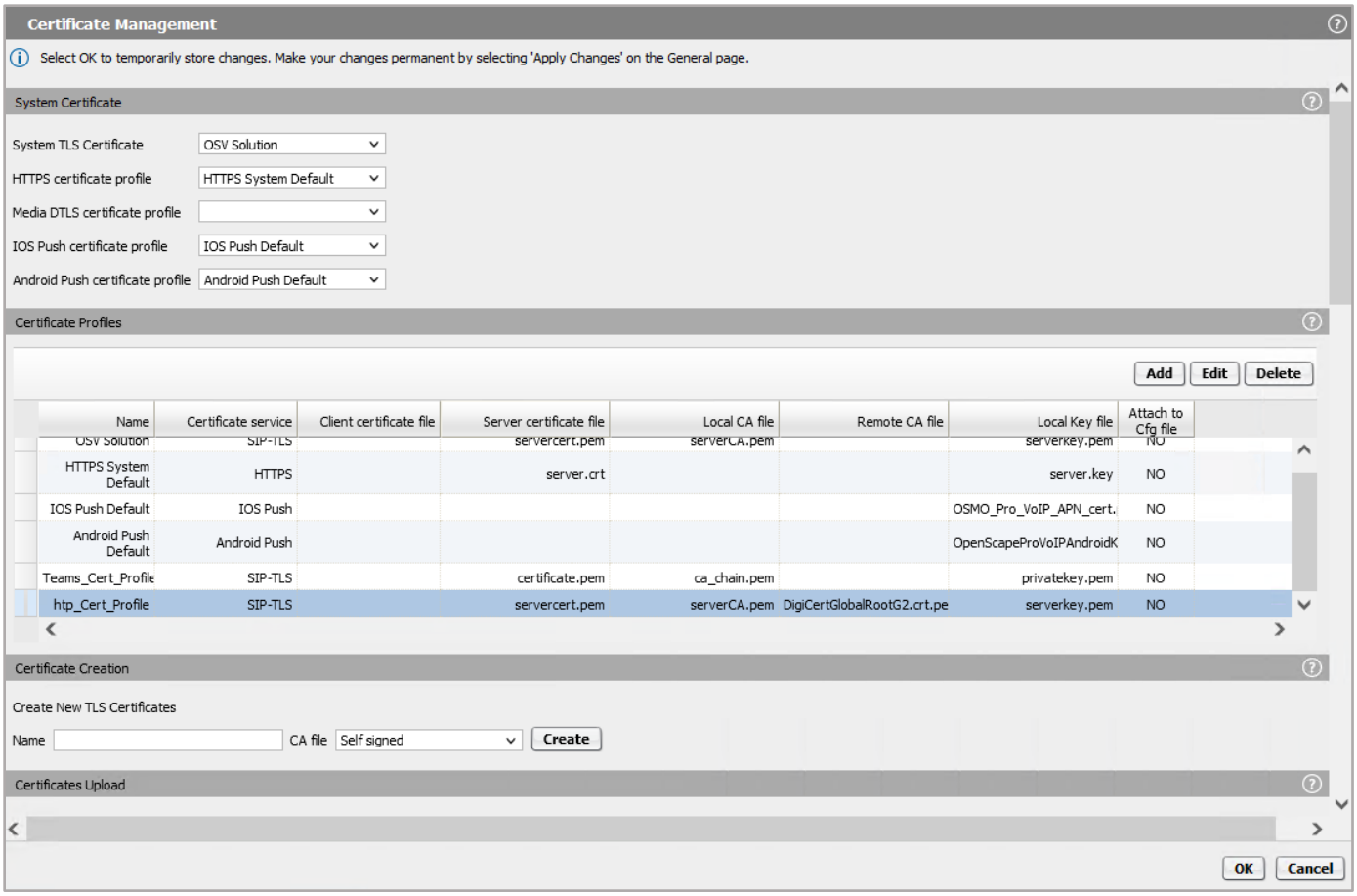
On the same window, click on **[Add]** to create the certificate profile.



Enter the following:

- **Certificate profile name**: htp_Cert_Profile (friendly name)
- **Certificate service**: SIP-TLS
- **Local server certificate file:** servercert.pem
- **Local CA file:** serverCA.pem
- **Local key file:** serverkey.pem
- **Remote CA file:** DigiCertGlobalRootG2.crt.pem
- **Minimum TLS version:** TLS V1.2

Click on **[OK]**.



Click on **[OK]** on **Certificate Management** window and then click on **[OK]** on **Security** window.

Click on **[Apply Changes]** on OS SBC main page.

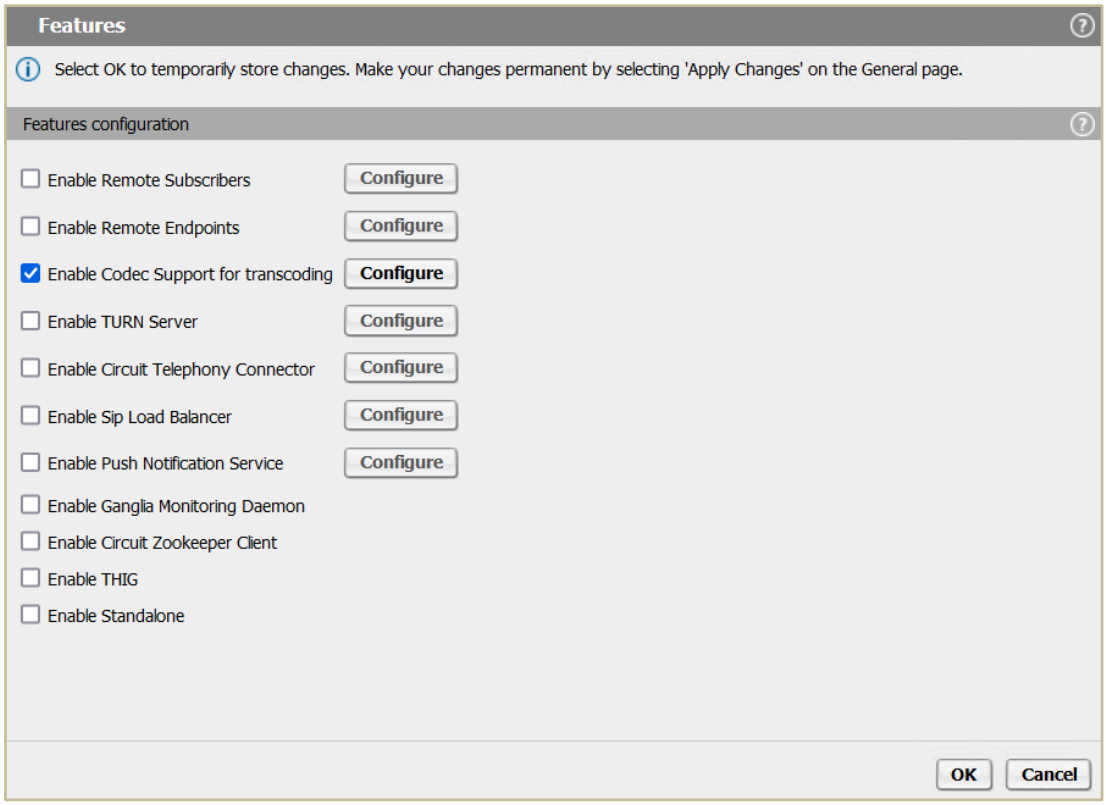
Media Profile

With **Media Profiles** settings, various parameters regarding the SDP messages and audio (RTP) traffic may be configured for the OS SBC SIP endpoints to SSP (htp) and OS4000.

Codec Manipulation Options

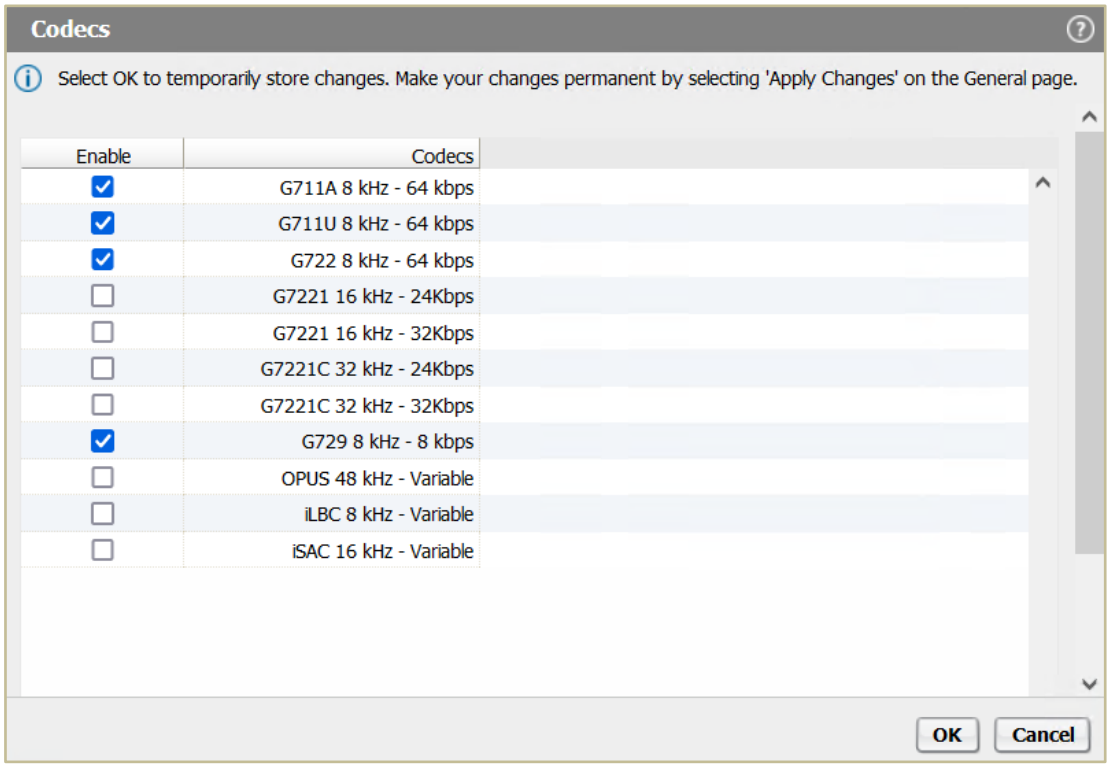
In case transcoding or certain codec prioritization for audio is required for the OS SBC – OS SBC – SSP media profile for the corresponding SIP trunk, it is required to enable the codec configuration options first for the media profile setup.

To do so, access **OS SBC Management Portal >> Features** window and select **"Enable Codec Support for transcoding"**.



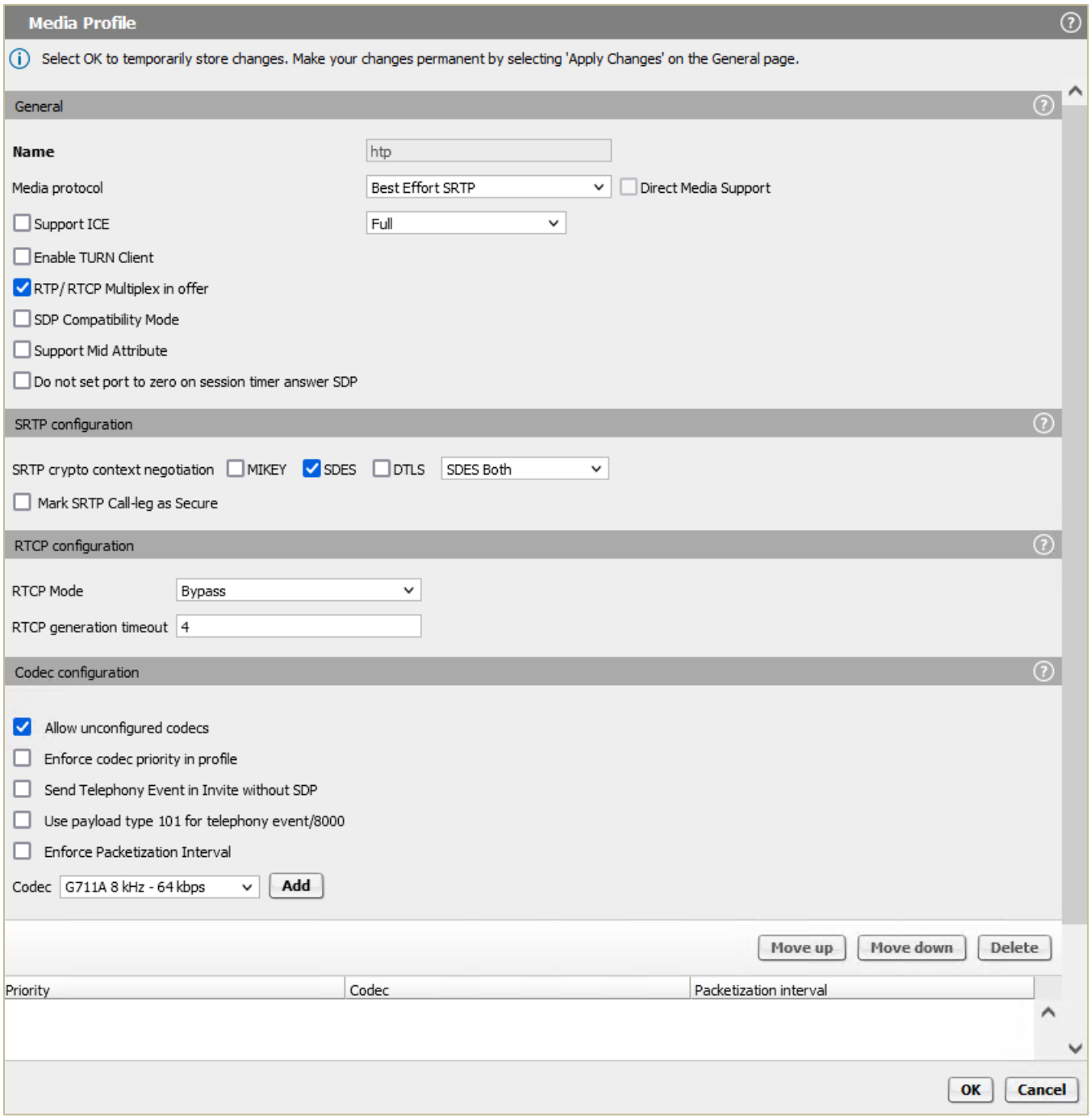
Click on **[Configure]**.

On **"Codecs"** window, select the codecs to be available for the media profiles (for e.g. transcoding, prioritization). As an example:



Click on **[OK]** and on the rest open windows.

Click on **[Apply Changes]** on OS SBC main page.



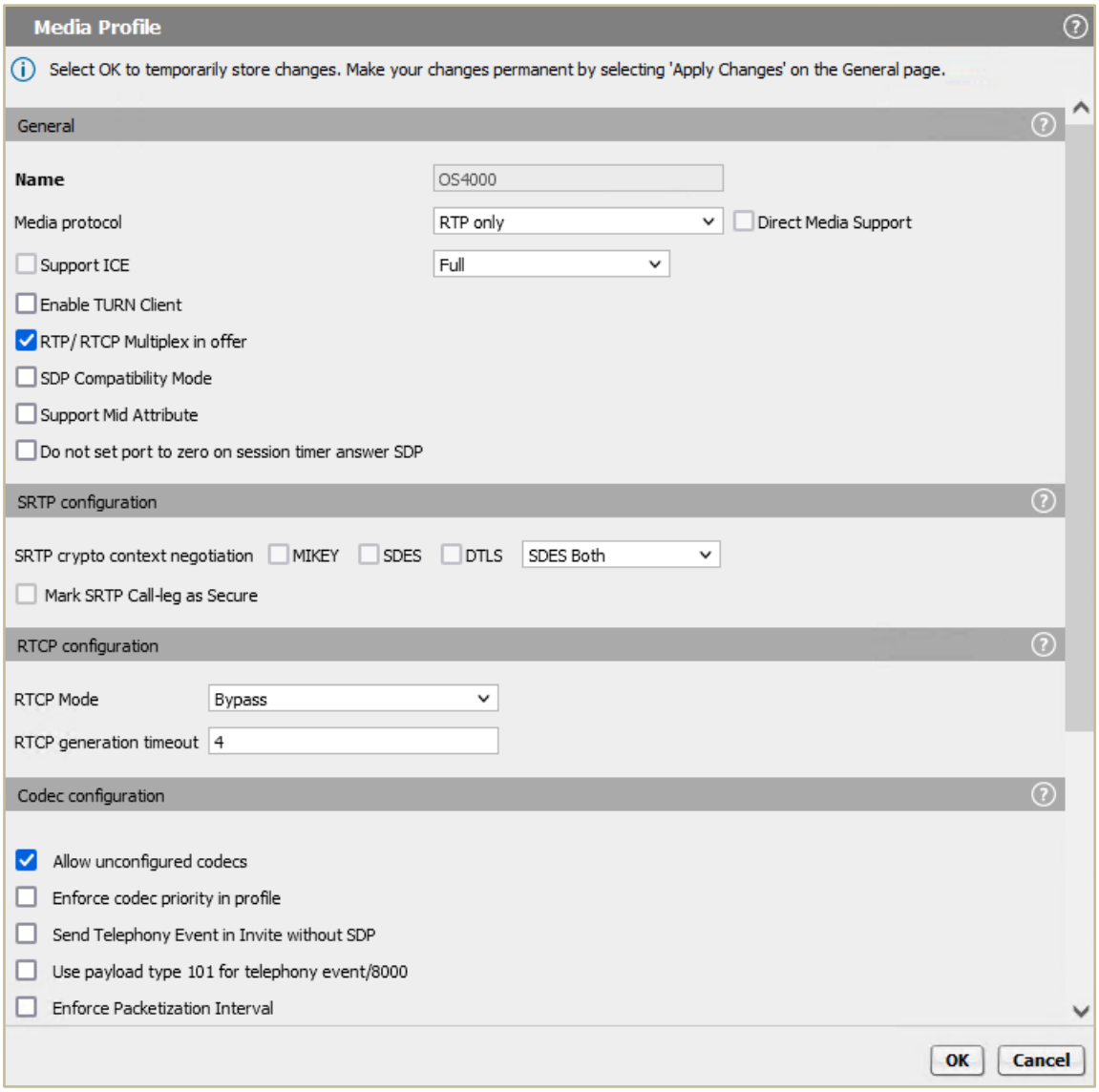
In **"Media Profiles"** area click on **[Add]** to create the media profile for OS SBC - SSP trunk by entering the following:

- **Name:** htp (friendly name)
- **Media protocol:** Best Effort SRTP
- **SRTP crypt context negotiation:** SDES (when secure media is required)
- **RTP/RTCP Multiplex in offer:** Enabled
- **Allow unconfigured codecs:** Enabled

Click on **[OK]** to return to **Media** window.

Click on **[OK]** on **"VOIP"** window.

Click on **[Apply Changes]** on OS SBC main page.



In **"Media Profiles"** area click on **[Add]** to create the media profile for OS SBC – OS4000 connection with the default settings:

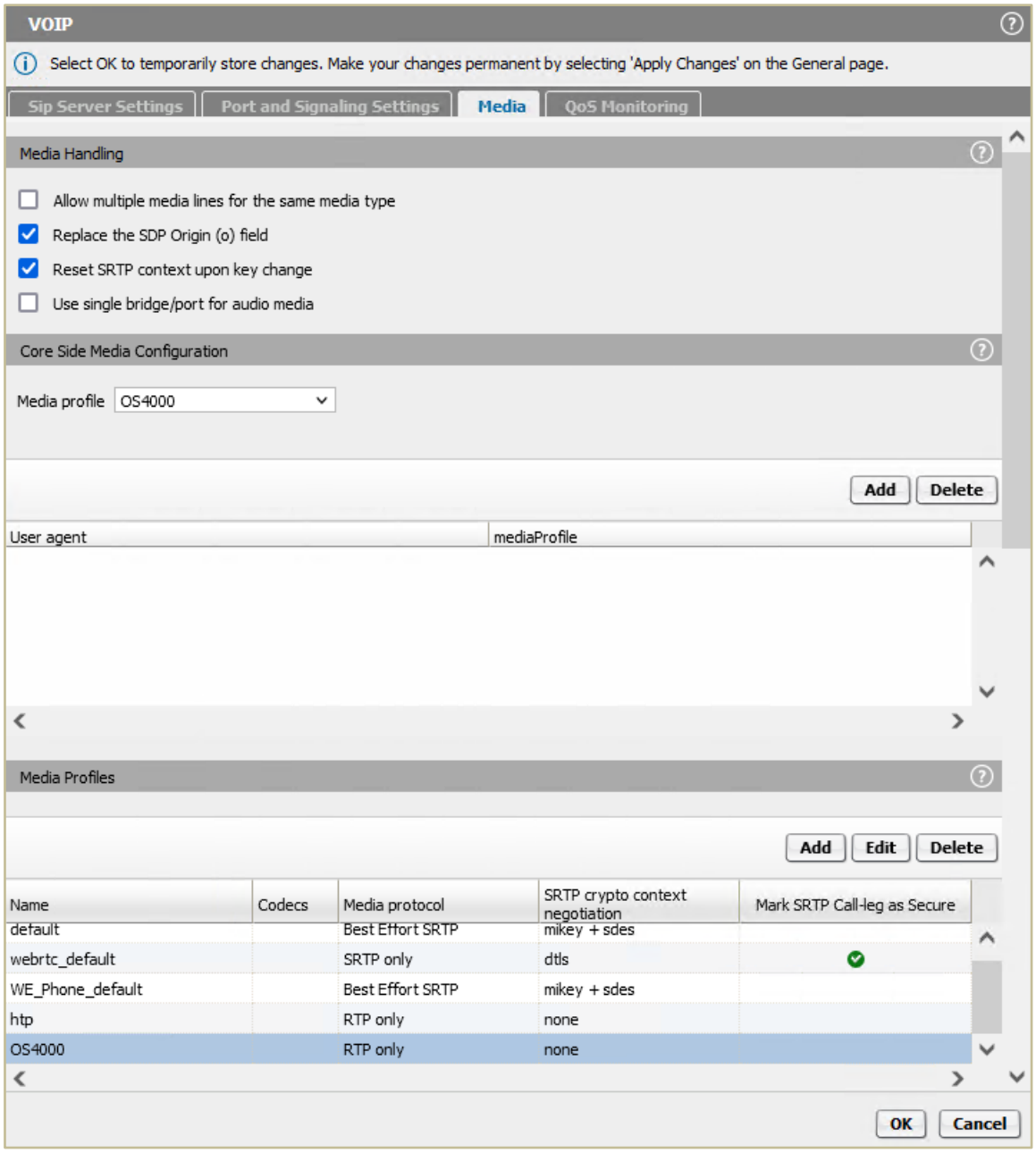
- **Name:** OS4000 (friendly name)
- **Media protocol:** RTP only
- **RTP/RTCP Multiplex in offer:** Enabled
- **Allow unconfigured codecs:** Enabled

Click on **[OK]** to return to **"Media"** window.

Click on **[OK]** on **"VOIP"** window.

Click on **[Apply Changes]** on OS SBC main page.

After creating the media profiles, configure the general media setting on **OS SBC Management Portal >> VOIP >> Media** window.



In **"Core Side Media Configuration"** area set "OS4000" from the **"Media profile"** dropdown list for the media profile used for the OS SBC – OS4000 SIP trunk.

In **"Media Handling"** enable the following:

- **Replace the SDP Origin (o) field:** Enabled
- **Reset SRTP context upon key change:** Enabled

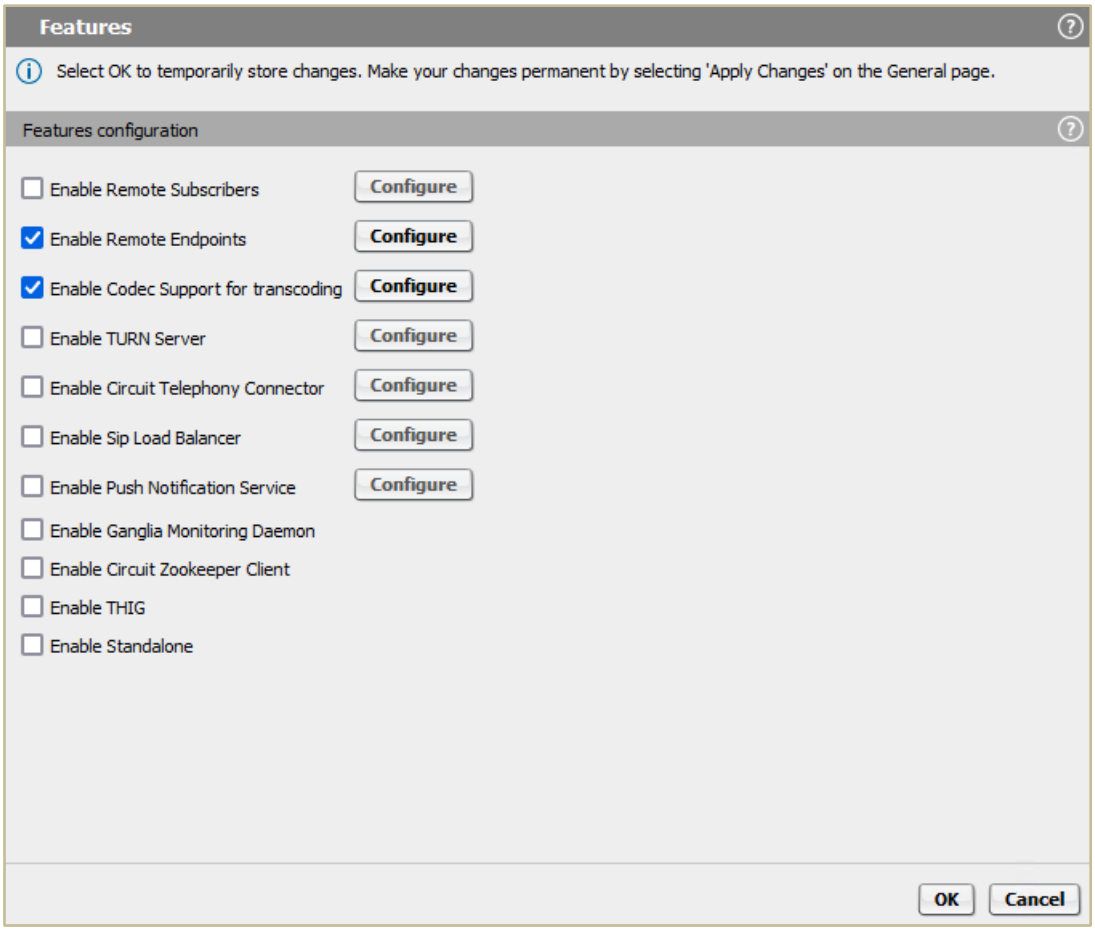
Click on **[OK]**.

Click on **[Apply Changes]** on OS SBC main page.

Remote Endpoints

In **Remote Endpoint** configuration the OS SBC with the SSP (htp) is setup.

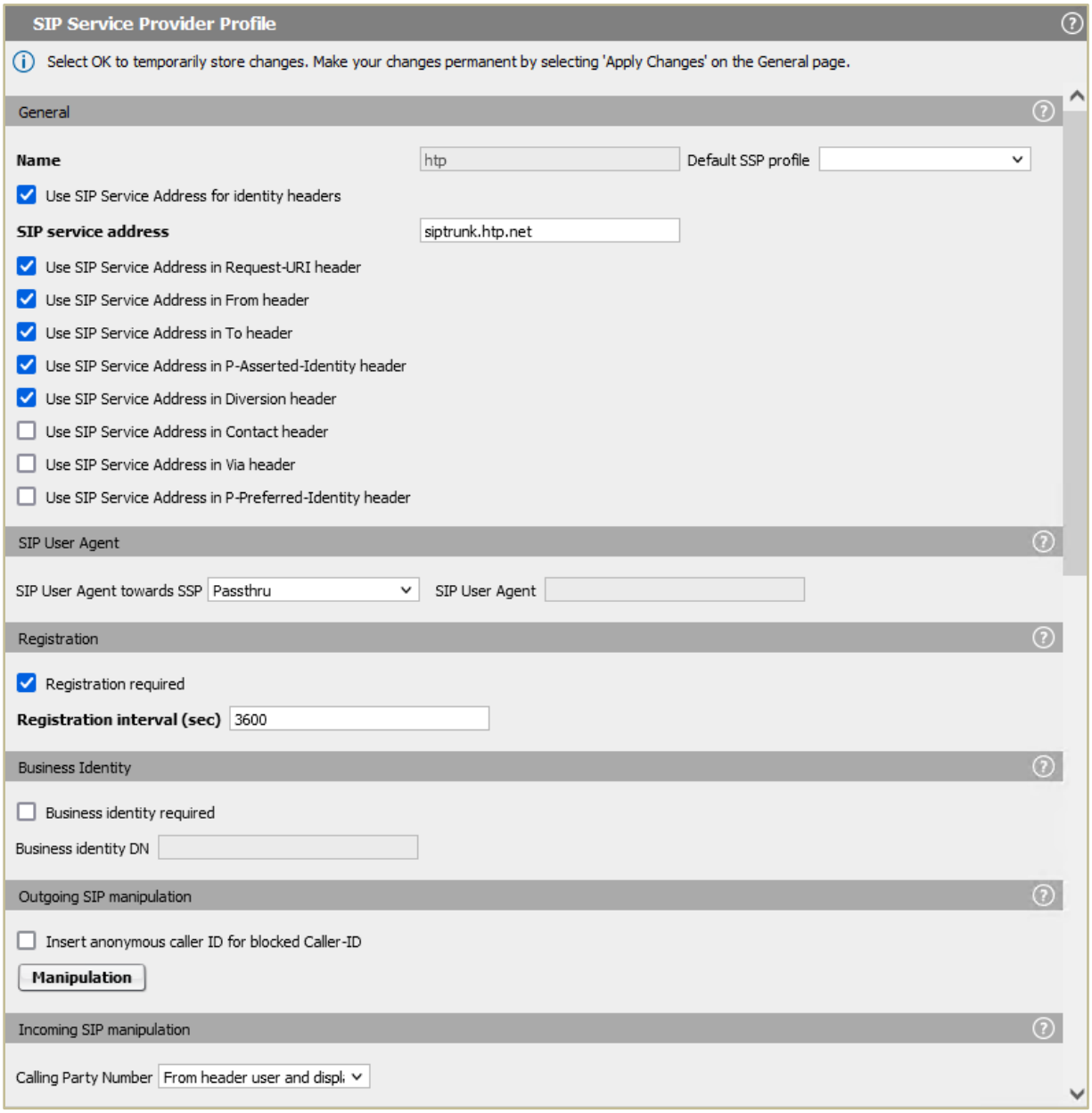
Navigate to **OS SBC Management Portal >> Features** window and set **"Enable Remote Endpoints"** to "Enabled".

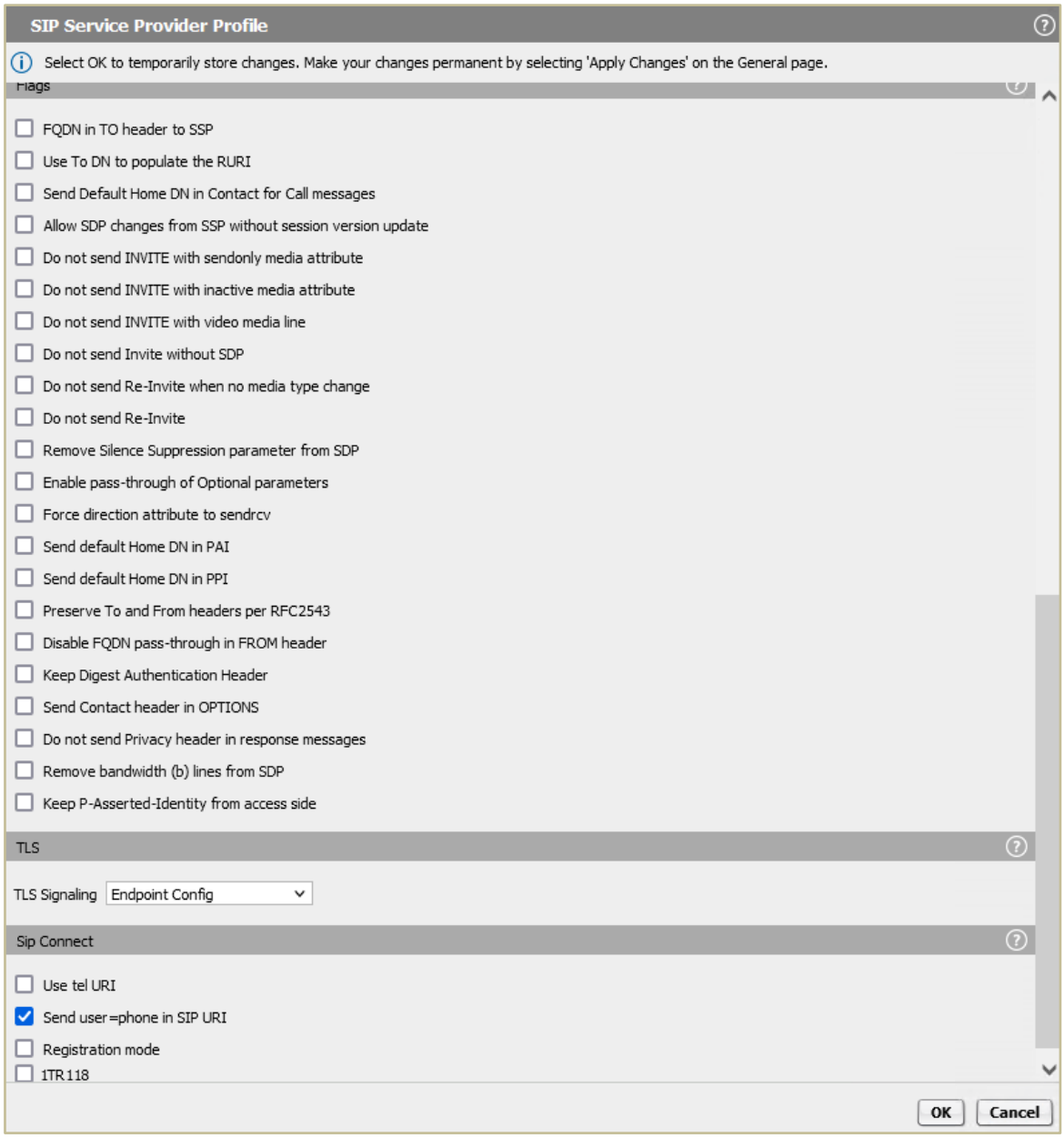


On **OS SBC Management Portal >> Features >> Enable Remote Endpoints** window, click on **[Configure]**.

SIP Provider Profile

On **Remote Endpoints** window click on **[Add]** in **"SIP Service Provider Profile"** area to add the endpoint profile for the OS SBC – htp SIP trunking.





On **"SIP Service Provider"** window, enter the following:

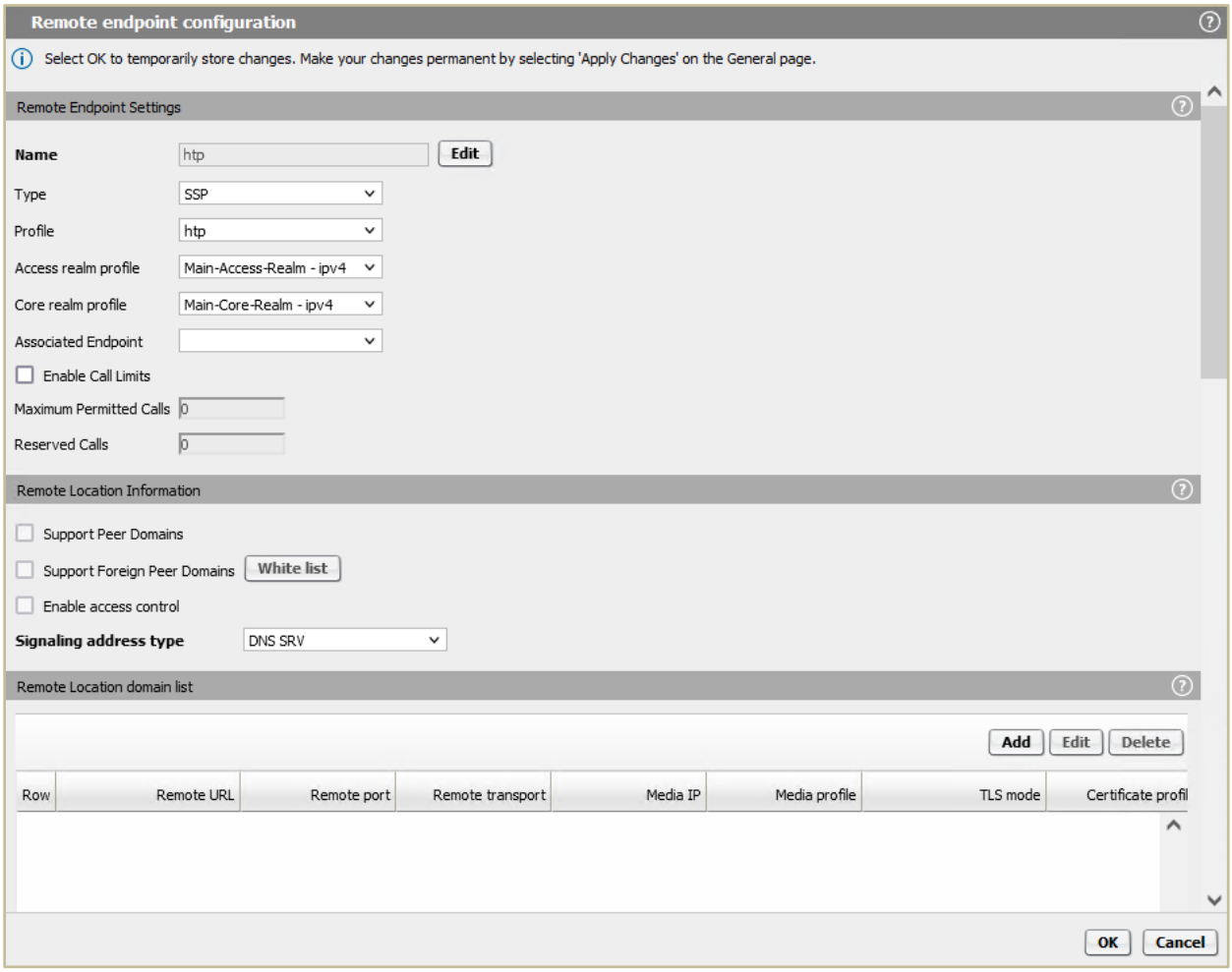
- **Name:** htp (friendly name)
- **Default SSP Profile:** Blank (in case the provider doesn't exist in the dropdown selection list, the field should remain blank and the user must manually configure the required flags for the SSP in use)
- **Use SIP Service Address for identity headers:** Enabled
- **SIP service address:** siptrunk.htp.net
- **Use SIP Service Address in Request-URI header:** Enabled
- **Use SIP Service Address in From header:** Enabled
- **Use SIP Service Address in To header:** Enabled
- **Use SIP Service Address in P-Asserted-Identity header:** Enabled
- **Use SIP Service Address in Diversion header:** Enabled

- **Registration required:** Enabled
- **Send user=phone in SIP URI:** Enabled

Click on **[OK]** to return to **"Remote Endpoints"** window.

Note: An htp **"Default SSP Profile"** will be available for user selection in future OS SBC versions.

In **"Remote endpoint configuration"** area, click on **[Add]**.



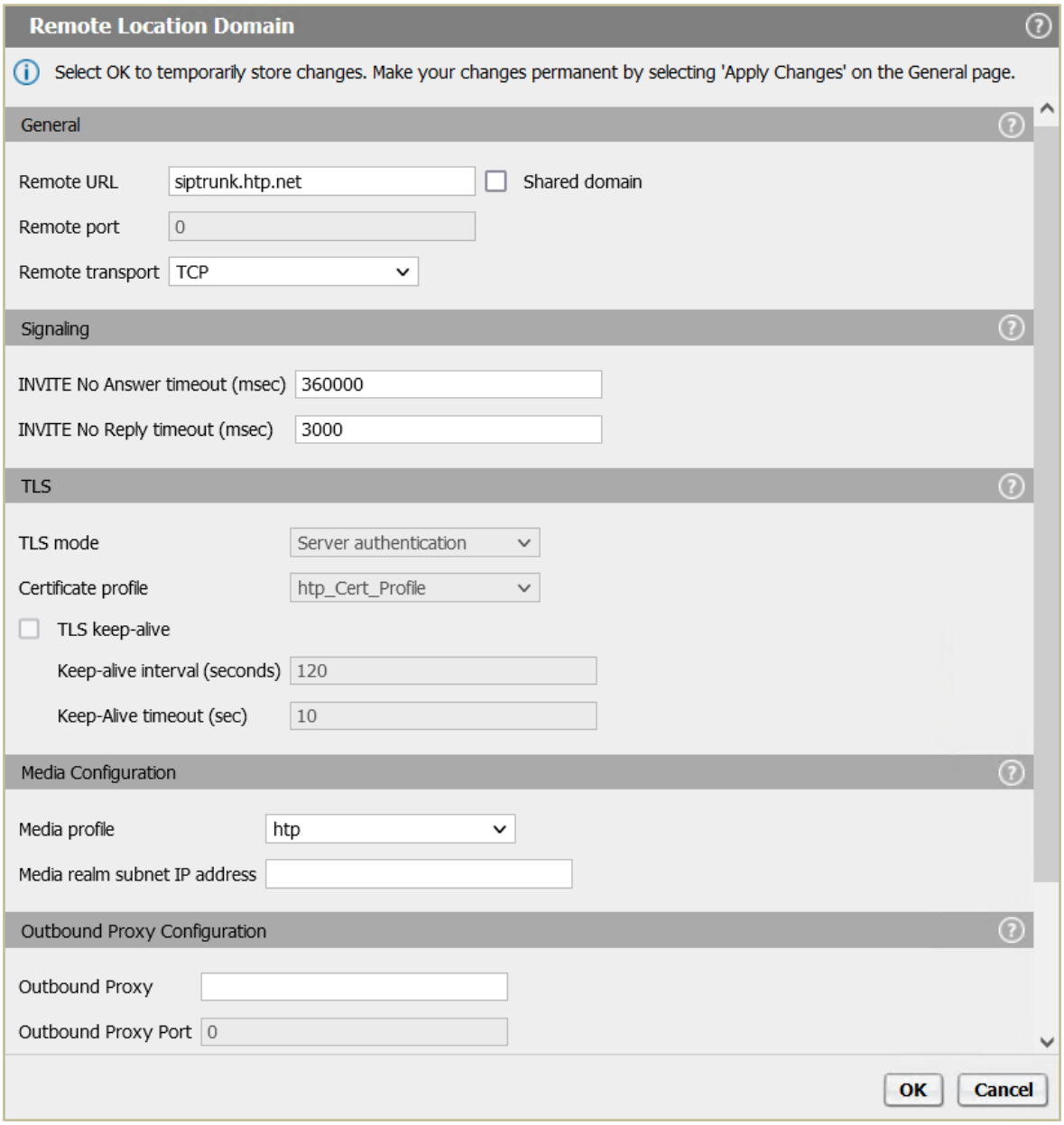
On **"Remote endpoint configuration"** window, enter the following in the **"Remote Endpoint Settings"** area:

- **Name:** htp (friendly name)
- **Type:** SSP
- **Profile:** htp

Continue to **"Remote Location Information"** area:

- **Signaling address type:** DNS SRV

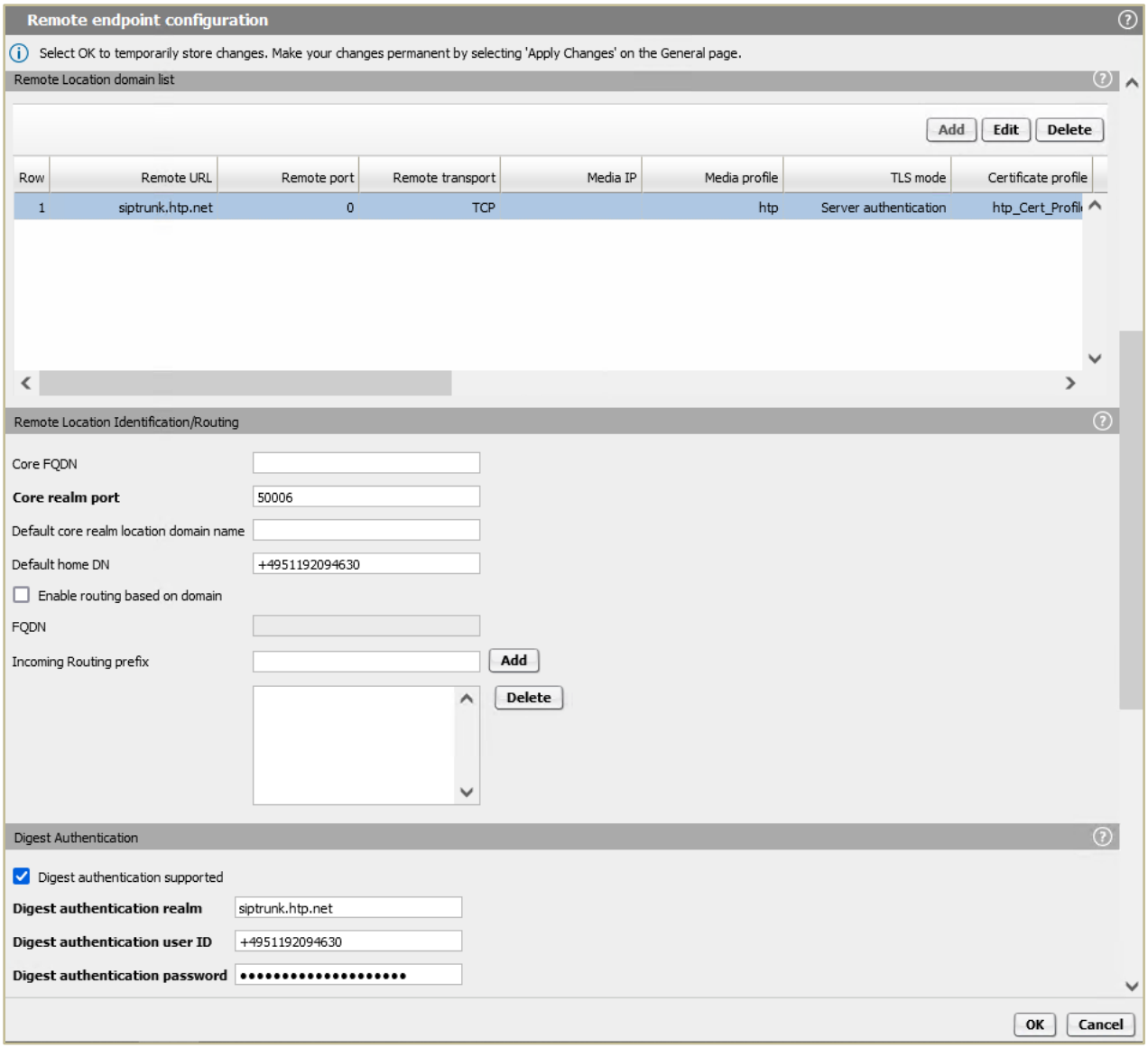
Click on **[Add]** in **"Remote Location domain list"** area.



On **"Remote Location Domain"** window, enter the following:

- **Remote URL:** siptrunk.htp.net (see section 4.1)
- **Remote port:** 0
- **Remote transport:** TCP (UDP or TLS are also possible)
- **TLS mode:** Server authentication (when TLS is used)
- **Certificate profile:** htp_Cert_Profile (when TLS is used- refer to sub-section 4.3.2)
- **Media profile:** htp (refer to sub-section 4.3.3)

Click on **[OK]** to return to **"Remote endpoint configuration"** window.



In **"Remote Location Identification/Routing"** area, enter the following:

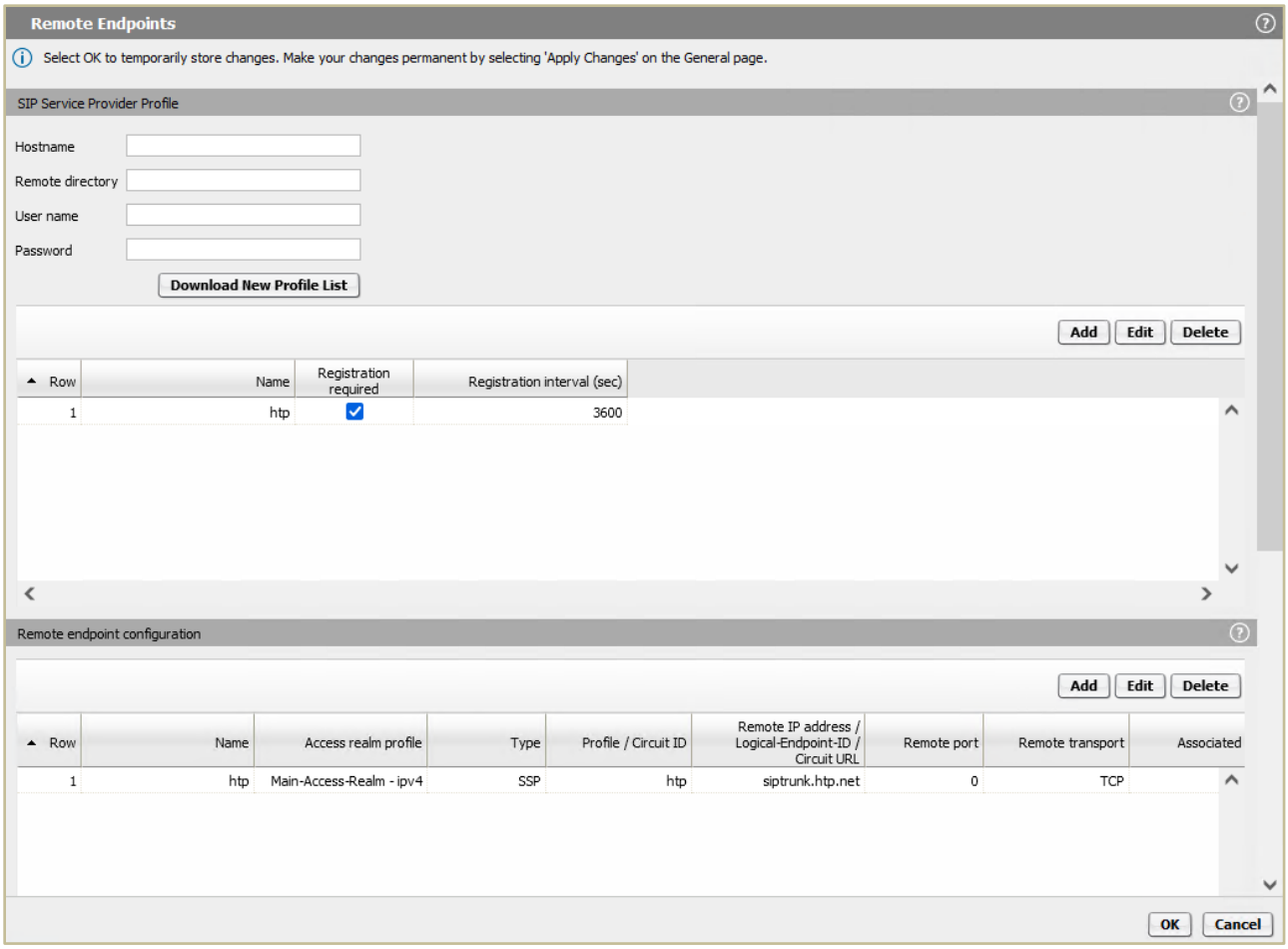
- **Core realm port:** 50006 (as configured in the OS$K WBM) 4.2.5)
- **Default home DN:** +49XXX

In **"Digest Authentication"** area, enable **"Digest authentication supported"** and fill in with the values below:

- **Digest authentication realm:** siptrunk.htp.net
- **Digest authentication user ID:** +49XXX
- **Digest authentication password:** *passphrase* (provided by htp)

Click on **[OK]** to return to **"Remote Endpoints"** window.

The **"Remote Endpoints"** window should look like the picture below:



Click **[OK]** on all open windows.

Click on **[Apply Changes]** on OS SBC main page.

Restrictions and Remarks

Remarks

- **General Information**

  In certification test environment there aren't any SPEP PSTN subscribers for testing, so SPPN (mostly) or SPGSM are used instead.

  The test scenarios were performed twice with both HFA and SIP OS4K internal stations.

  All call scenarios were executed with TCP transport protocol.

- **Voicemail**

  No voicemail service is provided by htp.

- **Conference**

  When OS4000 subscriber exits the conference, the PSTN subscribers continue to display the OS4000 subscriber (conference initiator), instead of each other.

  Conference display on various devices isn't propagated over the SIP trunk from OS4000.

- **Codecs**

  G.722, G.729 codecs aren't supported by htp's voice network.

- **DTMF**

  The htp network supports both RFC2833 and inband for DTMF transport methods. However, the inband isn't tested in current certification activities because htp is using only RFC2833 in production system and a change would affect all users.

  Atos Unify phone devices are using RFC2833 by default, but inband is also supported. There is no configuration option to disable RFC2833 on the phones. CP phones switch to inband only if remote side doesn't support RFC2833.

Restrictions

- **Basic Call**

  In a basic call scenario, where an OS4000 subscriber makes a call to a PSTN subscriber, the OS4000 subscriber displays only the number and not the name of the PSTN called party. Additionally, the PSTN subscriber displays only the number of the OS4000 calling party. On the other hand, when the PSTN subscriber is the calling party and the OS4000 subscriber is the called party, the PSTN subscriber will display the OS4000 subscriber's number and not the name. OS4000 sends PAI in *"200OK"* to htp, but this is not taken into consideration for IP phone display update.

  The htp service doesn't use PAI for display update purposes, but only for call validation, and display info is extracted by FROM header.

- **COLR**

  When a PSTN subscriber calls an OS4000 subscriber and the latter has COLR feature activated, OS4000 sends *"Privacy: id"* header in *"200OK"* to htp, but PSTN subscriber continues to display OS4000 subscriber's number. When htp receives *"Privacy: id"* header, the PAI header is removed from *"200OK"* by htp system, but this manipulation seems not to be enough for PSTN subscriber to have "Anonymous" call display. When the same scenario is tested with a networked OS Voice PBX node to OS4000, the caller has "Anonymous" call display.

  On the other hand, when the OS4000 subscriber is the caller and the PSTN subscriber on htp network has the COLR feature activated, there is no "Anonymous" call display on OS4000 subscriber. htp adaptation for COLR is not to send PAI, however OS4000 expects *"Privacy: id"* header in order OS4000 station to display an "Anonymous" call.

- **Call Hold**

  In a basic call between an OS4000 subscriber and a PSTN subscriber, when the OS4000 subscriber holds the call, the PSTN SIP subscriber doesn't display *"held remotely"* indication and hears MOH from OpenScape 4000. OS4000 sends re-INVITE (or ACK) for the hold with *"sendonly"* attribute in SDP.

  If the PSTN subscriber puts a call with an OS4000 subscriber on hold, the OS4000 subscriber won't display *"held remotely"* indication and hears MOH coming from PSTN. PSTN replies with an SDP attribute *"inactive"* in 200 OK. PSTN doesn't send SDP attribute "sendonly" or "inactive" in a hold re-INVITE.

- **Call Transfer**

  In call transfer scenarios (A->B->C), when subscriber "B" (*"transferring"* party) belongs to a different system than "A" (*"transferred"* party) and "C" (*"transferred to"* party) i.e., PSTN carrier vs OS4000, then after transfer is completed from "B", "A" and "C" will continue to display "B", instead of the connected party, when "A" and "C" are PSTN subscribers and "B" is an OS4000 internal station.

  After transfer completion, the display of the connected party information on the *"transferred"* and on the *"transferred to"* call participants, varies according to the relation with the *"transferring"* party (system-wise), if the *"transferring"* party is "A", "B" or "C" (in the indicative transfer scenario A->B->C mentioned above) and from the transfer flavor (attended, semi-attended, blind).

  The root cause for the display behavior in transfers is described in **"Restrictions – Basic Call"** bullet point. For more details refer to the corresponding test results in chapter 3.

- **Call Forwarding**

  In the indicative call forwarding scenario (A->B->C), when subscriber "B" (*"forwarding"* party) belongs to a different system than "A" (*"forwarded"* party) and regardless of where "C" (*"forwarded to"* party) belongs i.e., PSTN carrier vs OS4000, then after call forwarding completion from "B", "A" will continue to display "B", instead of the connected party and will not display forwarded call information. The same goes for various call forwarding

combinations in terms where the subscriber is registered and what flavor of call forwarding is used (Call Forward Unconditional, Call Forward Busy, Call Forward No Reply). Subscriber "A" will display the connected party only if "A" and "B" belong in the same system. Refer to test results in chapter 3 for the details.

- **Fax**

  The htp network doesn’t support fax exchange with T.38, but only with G.711.

Known Issues

- **Call Transfer**

  When an OS4000 SIP subscriber performs an attended transfer of a PSTN call to an OS4000 HFA subscriber, on HFA phone the connected party name isn't updated while the number display is correctly updated. The issue isn’t reproducible when the OS4000 HFA subscriber performs the transfer.