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Digital recorders KSRC series

CTI integration TRX call recorders with OpenScape Voice (HiPath 8000)

1:   Introduction

1.1   Scope

The document provides information about CTI integration TRX call recorders with Unify **OpenScape Voice** (formerly Siemens *HiPath 8000*) used for active recording calls within the system.

Our solution is compatible with the following IP phones:
- OpenStage 15,
- OpenStage 20,
- OpenStage 20 E,
- OpenStage 20 G,
- OpenStage 40,
- OpenStage 40 G,
- OpenStage 60,
- OpenStage 60 G,
- OpenStage 80,
- OpenStage 80 G,
- optiPoint 410,
- optiPoint 420

and provides two types of recording:
- automatic - every call from the monitored extension is being recorded
- on-demand - user can decide which call on monitored extension will be recorded at the touch of a key

Presently TRX solution support only static channel allocation. Therefore, the number of monitored extensions is equal to the number of recorder SIP endpoints (i.e. recorder channels) dedicated to recording calls within OpenScape Voice system.

1.2   Active recording

TRX integration is based on recorder SIP endpoints (registered in OpenScape Voice SIP registrar) and CTI link using CSTA (Computer-Supported Telecommunications Applications) protocol.
TRX recorder receives all necessary call events (call setup, call clearing) and call details (call setup date and time, call duration, CPA and CPB directory numbers) via CSTA protocol.
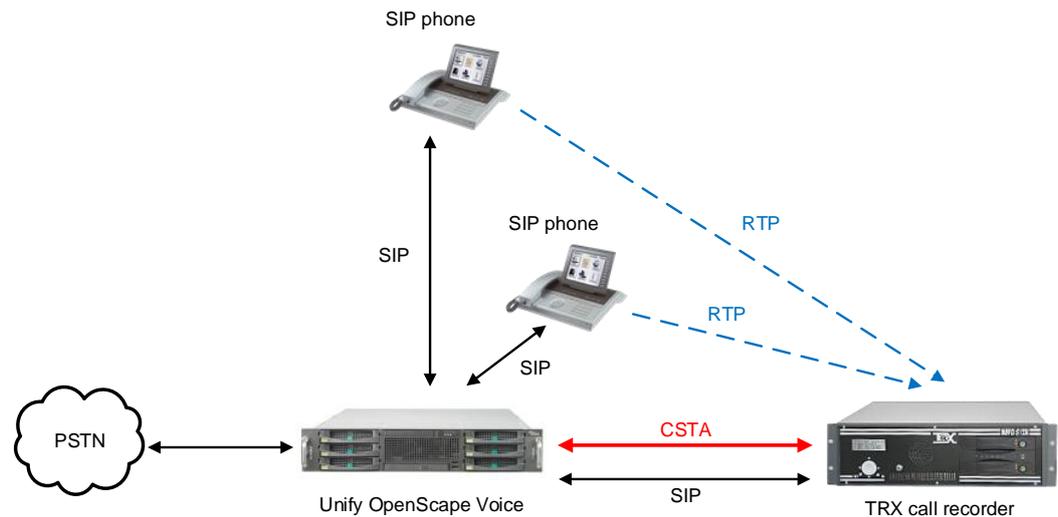
The audio stream (RTP) call participants is mirrored by recorded extension and sent to the recorder SIP endpoint (recorder channel). Hereby TRX recorder can record the entire voice flow of two or more participants.

2:   Configuring TRX recorder

To configure TRX recorder open *TRX Konsola 2* application, select your recorder and click on *Configuration* button.
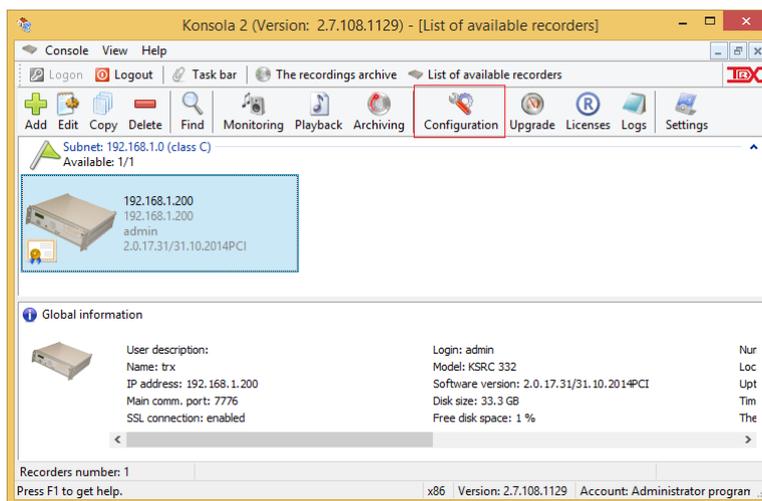
Next you should configure the following settings:
- CTI connection (link to CSTA server and recorder SIP endpoints),
- one or more virtual channels.

2.1   CTI connection

Click on *CTI* button, go to *Siemens, Multikom, Alcatel* bookmark and click *Manual configuration* link [1].
Enter or copy XML configuration to main window on the right. When you finish, click on *Apply* button [2].
Next send your configuration to the recorder. To do this, click on *Send* button [3] on top bar.
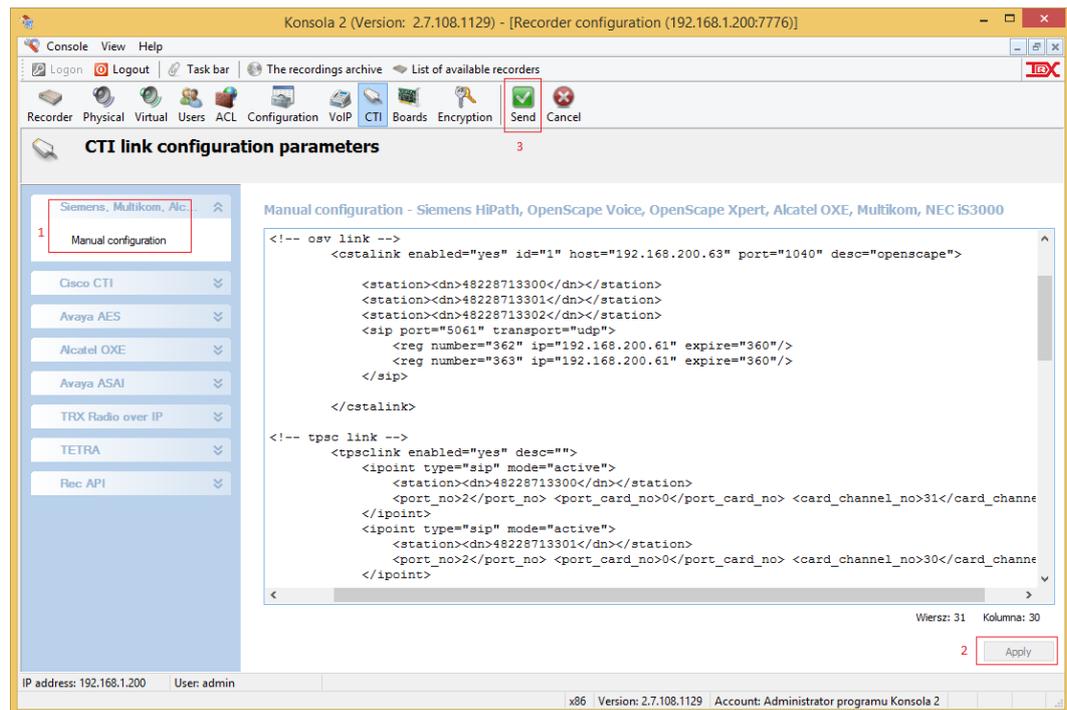
XML configuration data consist of three main sections (nodes):

cstalink – mandatory node – contains:
- IP address and TCP port number for CSTA server running on Open-Scape Voice;
- monitored (recorded) IP extensions directory numbers (DNs);
- IP address, transport layer protocol name and port number for SIP registrar.
  The SIP registrar is needed to register recorder SIP endpoints - used for active call recording on monitored extensions;
- Recorder SIP endpoints directory numbers (DNs).
  The passwords for recorder SIP endpoints are the same as their DNs;
- SIP session expire timeout (in seconds),
  After expiry of this time recorder SIP endpoint must renew registration;

tpsclink – mandatory node – contains data necessary to establish RTP links between OpenScape Voice and TRX recorder.
  tpsclink node consists of one or more *ipoint* nodes which determine recorder physical channels used to recording calls made/received from specified directory number.
  Recorder physical channel is determined by the three parameters: *port_no*, *port_card_no*, *port_channel_no*.

debug – optional node – used for debug purpose - reserved for TRX authorized services.

XML configuration sample:

```xml
<?xml version="1.0" encoding="utf-8"?>
<trxconfig>
    <cti>
        <systems>

            <!-- OSV -->
            <osv>

              <!-- csta link -->
                <cstalink enabled="yes" id="1" host="192.168.200.63" port="1040" desc="openscape">

                    <station><dn>48228713300</dn></station>
                    <station><dn>48228713301</dn></station>
                    <station><dn>48228713302</dn></station>

                    <sip port="5060" transport="udp">
                        <reg number="361" ip="192.168.200.61" expire="360" />
                    </sip>

                    <sip port="5061" transport="tcp">
                        <reg number="362" ip="192.168.200.61" expire="460" />
                        <reg number="363" ip="192.168.200.61" expire="460" />
                    </sip>

                </cstalink>

            <!-- TPSC links-->
                <tpsclink enabled="yes" desc="">

                    <ipoint type="sip" mode="active">
                        <station><dn>48228713300</dn></station>
                        <port_no>3</port_no> <port_card_no>0</port_card_no> <card_channel_no>31</card_channel_no>
                    </ipoint>

                    <ipoint type="sip" mode="active">
                        <station><dn>48228713301</dn></station>
                        <port_no>3</port_no> <port_card_no>0</port_card_no> <card_channel_no>30</card_channel_no>
                    </ipoint>

                    <ipoint type="sip" mode="active">
                        <station><dn>48228713302</dn></station>
                        <port_no>3</port_no> <port_card_no>0</port_card_no> <card_channel_no>29</card_channel_no>
                    </ipoint>

                </tpsclink>
```
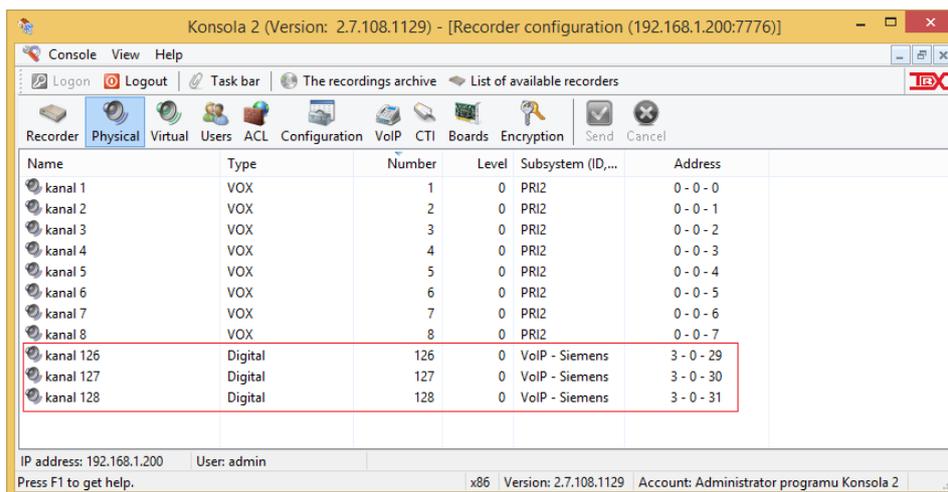
```xml
<!-- debug -->
    <debug>
        <app enabled="yes" user="yes" sys="yes">
            <solution />
            <config />
            <core />
            <links />
        </app>

        <tpsclinks enabled="yes" user="yes" sys="yes">
            <app />
            <rec />
            <tpsc />
        </tpsclinks>

        <cstalinks enabled="yes" user="yes" sys="yes">
            <csta />
            <cstaprv />
        </cstalinks>

        <sip enabled="yes" user="yes" sys="yes">
            <sep />
        </sip>

        <dispatcher enabled="yes" user="yes" sys="yes">
            <all />
        </dispatcher>

    </debug>
    </osv>
    </systems>
    </cti>
</trxconfig>
```

2.2   Physical channels

After completing CTI configuration go to the list of physical channels. To do this click on the **Physical** button.

If all settings are correct you can see new physical channels created in previous paragraph as recorder SIP endpoints.

The physical channels dedicated for call recording within OpenScape Voice should have **Address** value as you pointed in XML configuration data (*port_no - port_card_*no *- port_channel_no*).

The **Type** column should be *Digital*, and the **Subsystem** column - *VoIP – Siemens*.

You can change predefined names of this channels to user friendly names (up to 15 characters length).

2.3   Virtual channels

In the next step of configuration create one or more virtual channel. You can set up one virtual channel for all VoIP - Siemens physical channels, one virtual channel for every physical channel or any other combination.

To do this click on the **Virtual** button and run **Virtual channel wizard** from context menu.
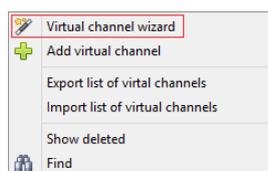
In subsequent steps of the wizard enter virtual channel name and assign physical channels to your virtual channel.
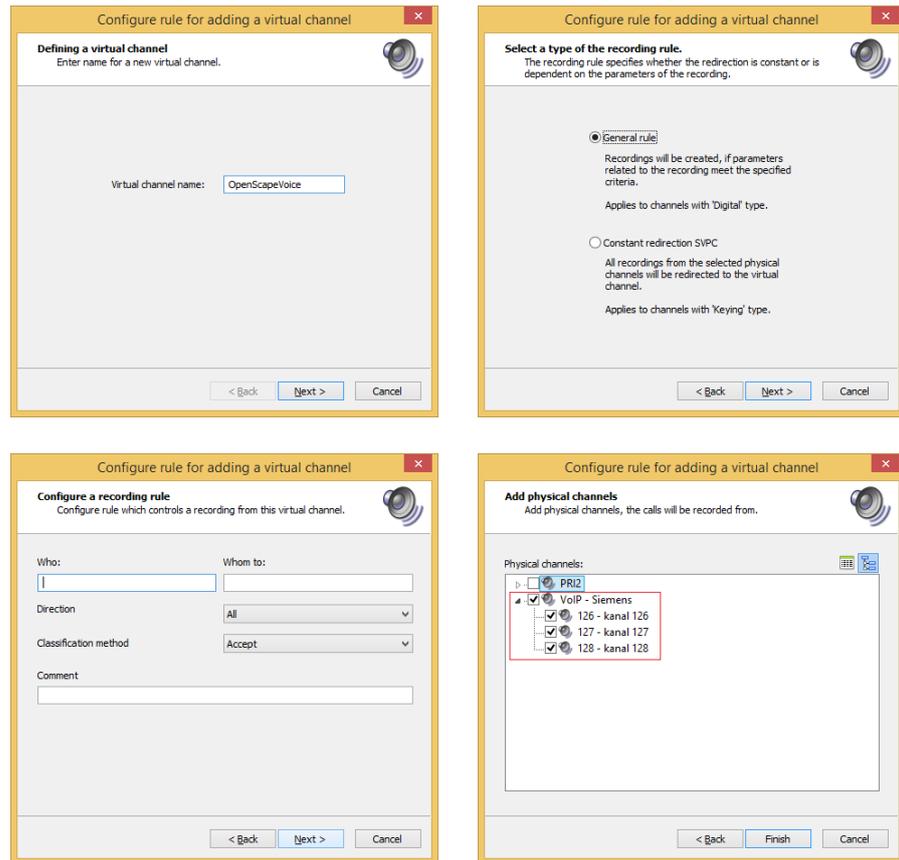
If parameters in **Configure rule for adding a virtual channel** window have the default settings, all calls will be recorded.

Within the **Who** and **Whom to** fields, it is possible define calling and called party numbers (or patterns) for calls, which are supposed to be recorded. The empty number field stands for any phone number. You can also allow for recording only incomming or only outgoing calls.

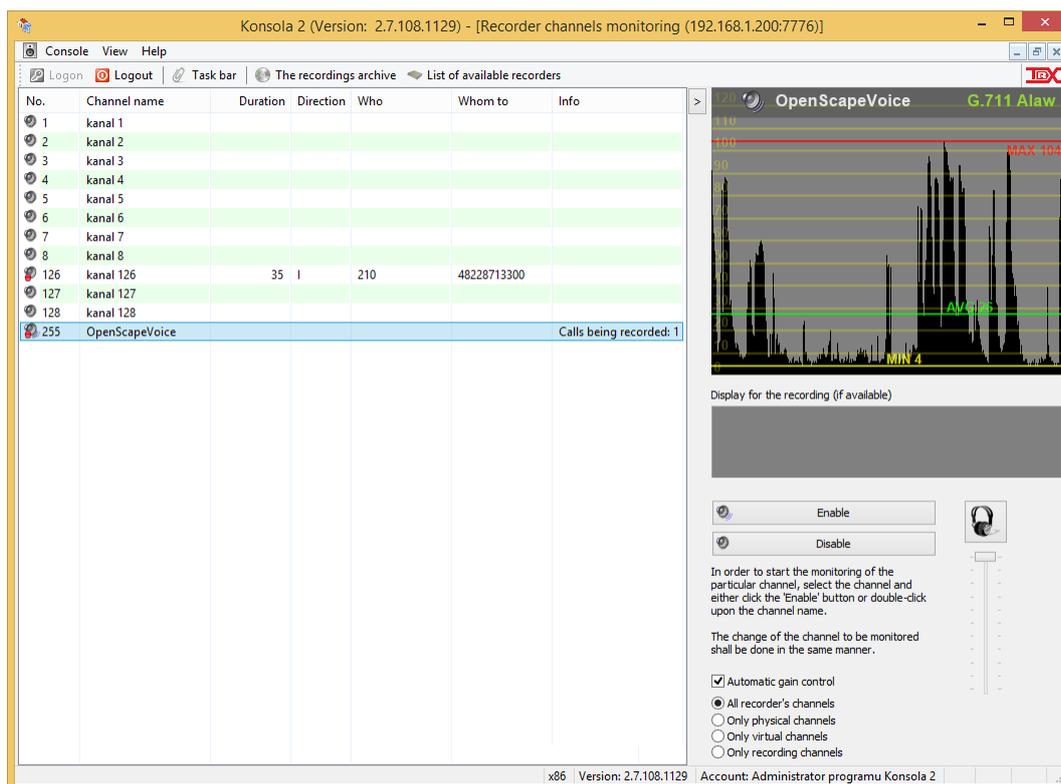For more details about virtual channel configuration see **Konsola 2 – user's manual** on http://www.trx.com.pl/en/dokumenty.
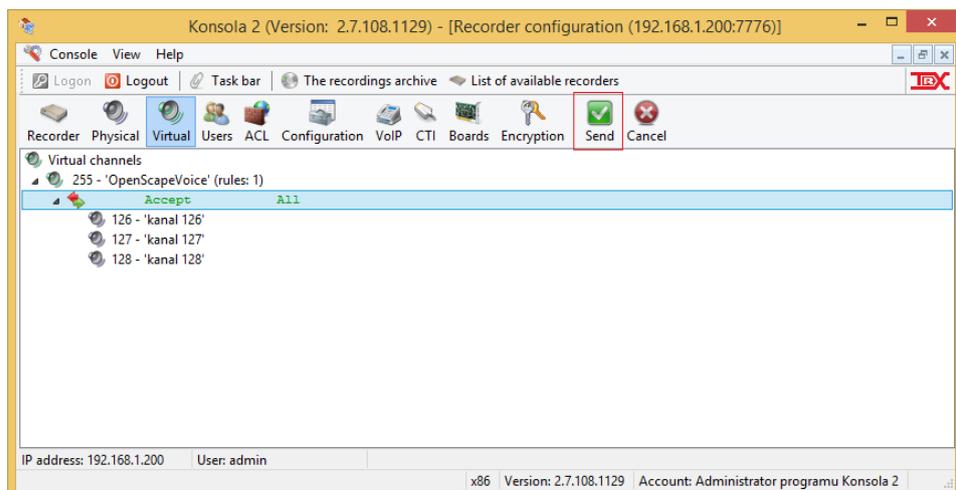
Next send your configuration to the recorder. To do this, click on *Send* button on top bar.

2.4 Channels monitoring

Recorder's channels status and eavesdropping is available right after the **Monitoring** module is launched. On the left side you can see all available channels (physical and virtual).

While a call is being recorded on the channel, the icon will change its appearance from 🔈 to 🔈 (there will appear a red dot on the speaker symbol). You can also see the current duration of monitored call and directory numbers of participants: **Who** and **Whom to** (only if recorder was able to obtain these data).

To enable monitoring a call on selected recorder's channel double click on the channel name or press the **Enable** button on the right side of the **Monitoring** module window. After a short time required for audio buffers to fulfill, the current conversation from the selected channel will be heard in the PC workstation sound system. The user can change the channel which he listens to at any time, in the ways described above.

The slider placed on the right side of the module window present the volume level of the conversation that is being recorded on the recorder. This level cannot be altered; however, it is possible to adjust the level of eavesdropped audio signal which is reproduced by the local computer's sound system.

Automatic gain control checkbox allows you to automatically control audio level of eavesdropped calls.

Automatic gain control function does not change the parameters of the original recording that is saved on recorder hard disk.

2.5   Playlist

In order to gather a list of recordings from the recorder, it is necessary to specify the following parameters:
- **source of the recordings**, i.e. logical and / or physical channels,
- **time range** for searched calls.

The *Playback* window provide also browsing and filtering list of calls. There is a possibility to add text notes or attachments (files of up to 2 MB) to the selected recording. Column of the table can be can be shown and hidden and placed in any order

Any calls from the list can be saved to audio files (*wav* or *mp3*) or TRX proprietary archive (tra3). Selected records can be saved to text files in *csv* or *html* format.

Every user (created on TRX recorder) can see only calls recorded on channels, which he has granted access to.

TRX solution can enhance CDR by three additional parameters:

Calling name	- calling party name,
Answering device number	- number of the extension that has pickup the call (if the call was forwarded or transferred - this number may be different than number placed in *Called* column),
Answering device name	- name of the extension that has pickup the call

2.6   Player

The *Player* window allows to playback the selected recording and provides additional information relating to the recording. It is divided into a few sections, presenting different information related to the recording.
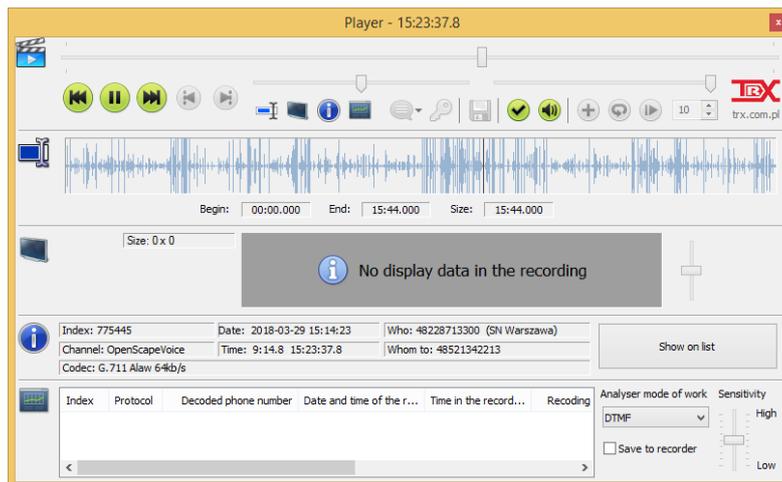
The top slider indicates the part-listening position relating to the entire call duration, slider also enables playback the call from the marked point of time.

The left bottom slider allows adjusting the speed of playback process (in the range -50% ... +50%). Position of this slider is remembered for **all** played recordings.

The ✔ button (next to the „floppy") restores natural playback speed.

The right bottom slider allows rice or lover the volume for played calls. The slider setting is stored for all recordings recorded on **the same channel** (even after restart the *Konsola 2* application).

The 🔊 button means inactive, and the 🔇 button - enabled *MUTE* function (while the playback continues).

The ➕ button in green color means that the *Automatic gain control* (*AGC*) function is active. This feature allows switching the mechanism causes all played calls have equal, like each other volume level.

Automatic gain control **does not change the volume of the original recordings stored on the recorder.**

The 💾 button allows saving whole or selected part of current recording to audio file (*wav* or *mp3*).

Group of buttons ⏮ ▶ ⏭ or ⏮ ⏸ ⏭ or (depending on whether the current recording is played, or not) enables appropriately:
- jump the playback to the previous call on the list,
- pause or play/resume current sample,
- jump the playback to the next record in the list.

🔁 [*Continuous playback*] button allows to switch on or switch off jumping to the next record.

⏯ [*Play beginnings (sec.)*] button allows to switch on or switch off playback only the be-ginning of recordings. Duration of the played parts of the calls can be determined in the range of 1 to 300 second.

3:   Configuring SIP extension

Every extension should be adding manually to the list of monitored and recorded stations (using appropriate key sequence in local menu).
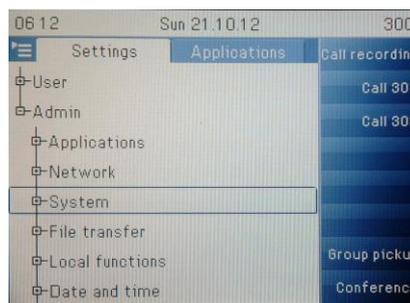
3.1   Call recording enabling

Enter to the menu and set up the following parameters:
- recording SIP endpoint directory number (recorder number)
- recording mode

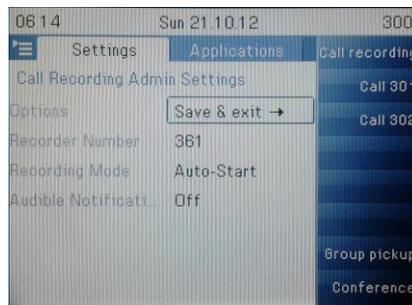To do this for most OpenStage series terminals go to the:
Settings > **Admin** > **System** > **Features** > **Configuration** > **Call Recording** menu path.

To configure automatic recording set parameter **Recording mode** to **Auto-start**.

Enter the recording SIP endpoint directory number in **Recorder Number** field.

Save settings and exit menu.

CAUTION:

Pressing the HOLD key turns of call recording.
Pressing the RETRIEVE key turns on call recording.

4: Configuring OpenScape Voice

4.1 SIP endpoints

Open OpenScape Voice control panel and for every required recorder SIP endpoint follow these steps:

- create SIP account
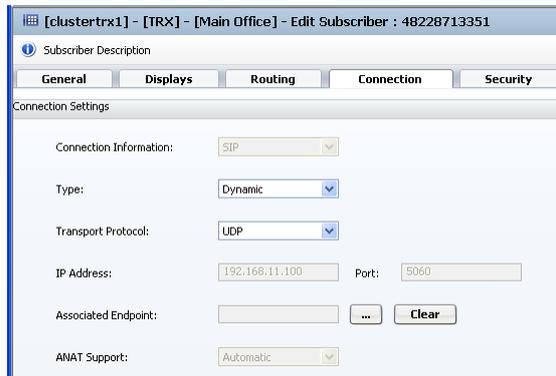
- set *Password* value the same as *User Name*
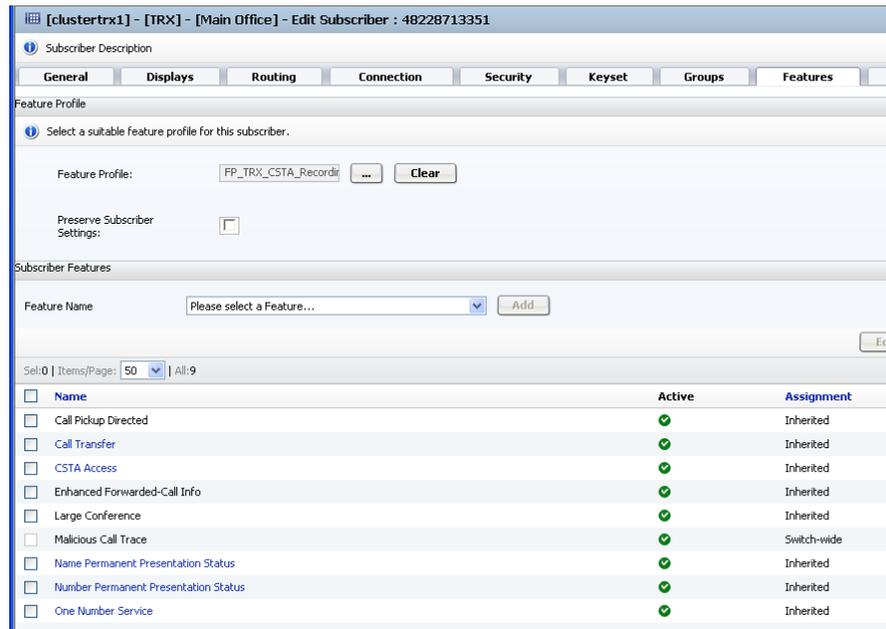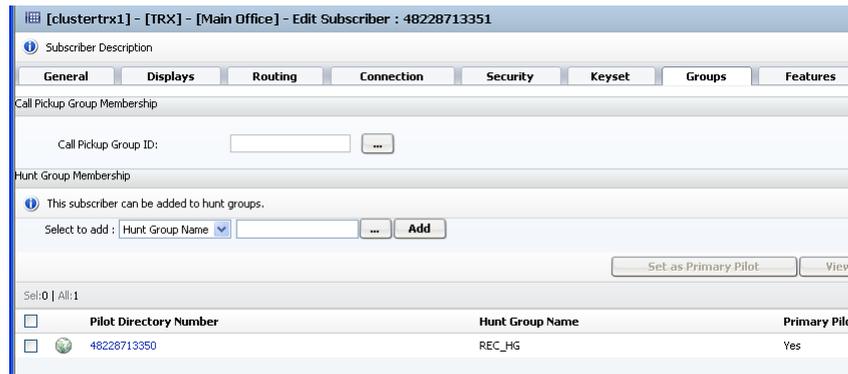
4.2 Hunt Groups

Create Hunt Group and add to the group all recorder SIP endpoints.
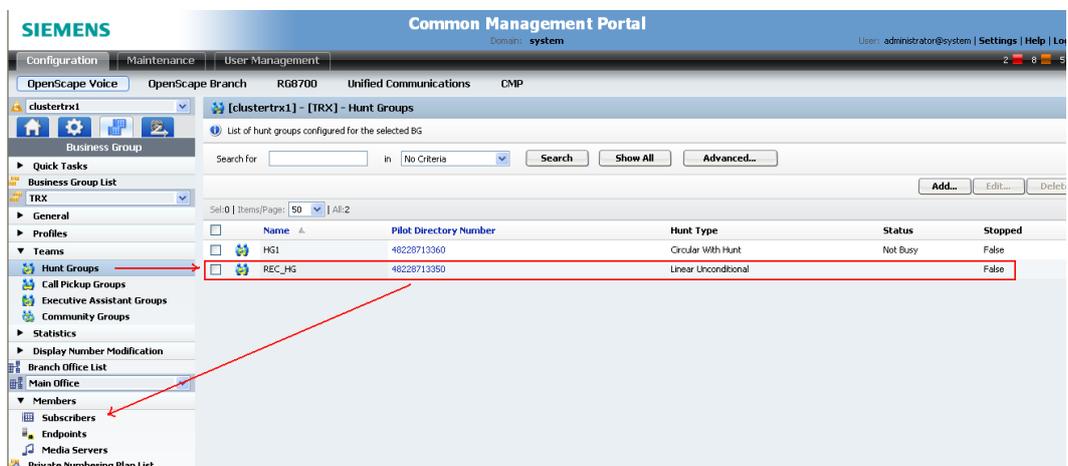
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