Why does my Snom phone always end the current call after a certain time

Problem

For no apparent reason **my Snom phone ends the current call after about 15 minutes**!

Solution

To Understand the issue let's take a look at the following SIP communication between **Snom Phone** and **PBX**:

- The Snom Phone with number **123456** sends an Invite to **7890**;
- The call comes in and is established between **123456** and **7890**, so far so good.

> **Note: For better overview we have removed some SIP headers from the communication and highlighted the most important ones in this error analysis.**

```
Sent to Udp:217.x.xx.xx:5060 from Udp:192.168.2.124:50422 at Jan 30 12:57:23.743 (1419 bytes):
INVITE sip:7890@testpbx.de;user=phone SIP/2.0
From: "Test" <sip:123456@testpbx.de>;tag=dzadbqiwhi
To: <sip:7890@testpbx.de;user=phone>
Call-ID: 3135383033383534343035313335 30-mxeagr7ulapk
CSeq: 2 INVITE
User-Agent: snomD765/10.1.49.11
Contact: <sip:123456@192.168.2.124:50422>;reg-id=1
X-Serialnumber: 000413xxxxxx

Supported: timer, 100rel, replaces, from-change
Accept: application/sdp
Session-Expires: 3600
Min-SE: 90
------------------------

Received from Udp:217.x.xx.xx:5060 on Udp:192.168.2.124:50422 at Jan 30 12:57:28.017 (1339 bytes):

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.124:50422;received=84.141.56.123;rport=50422;branch=z9hG4bK-qa5k4ddrhvok
To: <sip:7890@testpbx.de;user=phone>;tag=h7g4Esbg_p65542t1580385444m252622c553395349s1_688083261-1877431650
From: "Test" <sip:123456@testpbx.de>;tag=dzadbqiwhi
Call-ID: 313538303338353434303531333530-mxeagr7ulapk
CSeq: 2 INVITE
Contact: <sip:partner@217.x.xx.xx;transport=udp>
Session-Expires: 1800;refresher=uas
Supported: timer
```

In this case it is very important what is negotiated between the Snom phone and the other party (PBX server).

- Let us concentrate on the parameters that are important for the duration of the session (call).
- The Snom phone suggests a session duration of 3600 seconds; see: **Session Expires: 3600**
- In the **200 OK** the PBX changes this value to Session **Expires: 1800** (=30 minutes).
- Additionally it defines who renews the session: **refresher=uas. This means that the (uas: user agent server) takes over the session refresh. In this case the PBX.**

What does the RFC say: The refresher has to send an update after the half of the time. In our case it is exactly 15 minutes.

- If the client (in this case the Snom phone) does not receive any message from the server after half the time to keep the session running, the client terminates the session.
- What the Snom phone also does with the sending of the **BYE**.
- This procedure is described here: https://tools.ietf.org/html/rfc4028

Here you can see the BYE being sent by the phone after 15 minutes and 13 seconds:

```
Sent to Udp:217.x.xx.xx:5060 from Udp:192.168.2.124:50422 at Jan 30 13:12:41.714 (699 bytes):

BYE sip:partner@217.x.xx.xx;transport=udp SIP/2.0
From: "Test" <sip:123456@testpbx.de>;tag=dzadbqiwhi
To: <sip:7890@testpbx.de;user=phone>;tag=h7g4Esbg_p65542t1580385444m252622c553395349s1_688083261-1877431650
Call-ID: 3135383033383534343035313335330-mxeagr7ulapk
CSeq: 4 BYE
User-Agent: snomD765/10.1.49.11
Contact: <sip:123456@192.168.2.124:50422>;reg-id=1
Content-Length: 0
```

In the SIP Trace we also see that right after the Snom phone sends the BYE, the PBX sends an update for the session, but this comes late(13:12:43.583).

```
Received from Udp:217.x.xx.xx:5060 on Udp:192.168.2.124:50422 at Jan 30 13:12:43.583 (716 bytes):

UPDATE sip:123456@192.168.2.124:50422 SIP/2.0
To: "Test" <sip:123456@testpbx.de>;tag=dzadbqiwhi
From: <sip:7890@testpbx.de;user=phone>;tag=h7g4Esbg_p65542t1580385444m252622c553395349s1_688083261-1877431650
Call-ID: 3135383033383534343035313335330-mxeagr7ulapk
CSeq: 3 UPDATE
Contact: <sip:partner@217.x.xx.xx;transport=udp>
Min-Se: 900
Session-Expires: 1800;refresher=uac
Supported: timer
Content-Length: 0
```

Conclusion:

- This proves that the problem is on the server side, because it does not update the session in time.

Further Information

- VoIP Essentials
- Visit the Snom Forum
- Open a support ticket
- Find a local partner

Related articles

- Entering Snom Deskphone Web Interface - WUI
- Get the snom phone ready for operation
- How can I backup and restore the phone configuration
- How can I configure a second hand Snom device that still points to the old PBX because of a redirection
- How can I obtain a SIP trace from the phone
- How to - Core Dump
- How to capture large or permanent PCAP traces
- How to obtain a log from a desktop phone
- How to obtain a SIP trace from a deskphone
- How to setup a Syslog server on a Desktop phone
- How to store a PCAP trace into an USB mass-storage device
- How to store a phone syslog into an USB mass-storage device
- How to store a SIP trace into an USB mass-storage device

- Incoming calls are rejected by "403 Use Proxy"
- Is my Snom phone still covered by warranty
- My phone displays the SIP Disabled - error, how can I fix it
- Reboot your Snom Phone
- Setting up a Snom Deskphone
- Setting up WLAN Connection on Snom Deskphone
- Store HTTP-Trace on USB Stick
- Store Pcap on USB Stick
- Store SIP-Trace on USB Stick
- Store Syslog on USB Stick
- Troubleshooting How-to
- What kind of DTMF types Snom phones support and how to change them
- Why does my Snom phone always end the current call after a certain time