AudioCodes 5.80a Stand-Alone Survivability

Here are the configuration steps for a typical way to use the AudioCodes Stand-Alone Survivability (SAS) Feature with AudioCodes 5.80a firmware and later.

Introduction

The SAS feature can be useful for a branch office that has a local AudioCodes Gateway, but uses a remote sipXecs server. In "Normal" mode the AudioCodes Gateway transparently proxies the SIP messages between the phones and sipXecs server. It also sends periodic "keepalive" messages to the sipXecs server in order to confirm connectivity. If connectivity to the sipXecs server is lost then the AudioCodes Gateway goes into "Failover" mode allowing branch phones to make calls out the PSTN and to each other. Once connectivity is restored, the AudioCodes Gateway switches back to "Normal" mode.

The instructions below are based on the results of my testing with an AudioCodes Mediant 1000 PRI (6.00 firmware), an Avaya 1230, and an LG-Nortel 6810.

Configuration Steps

sipXconfig Phone Profiles

To have a phone use SAS, configure its Outbound Proxy values as follows:

- Address: The FQDN of the AudioCodes Gateway, e.g. gw.example.com
- Port: 5080 (yes 5080, not 5060)

This forces the phone to send its all SIP messages through the AudioCodes Gateway.

Phones that do not have a configurable Outbound Proxy cannot make use of the AudioCodes SAS feature. (Well, at least not using these instructions.)

Be aware that in sipXecs 4.2 the Polycom SoundPoint IP profiles by default include per-Line Outbound Proxy values, found under "Lines" - X - "Registration" (and "Show Advanced Settings".) The per-Line values take precedence over the per-Phone values found on the "SIP Servers" screen. You need to change the per-Line values.

The profile of course also needs to be re-generated and sent to the phone.

sipXconfig Dial Plan

The Dial Plan must be constructed so that all digits dialed by the user are passed to the AudioCodes Gateway. i.e. None of your PSTN rules can have prefixes that are dropped before the call is sent to the gateway.

This means you cannot use the built-in "Local" or "Long Distance" rules, and you will need to construct "Custom" rules instead. The "Emergency" rule may be used, although you should test that it works in both Normal and Emergency mode.

sipXconfig AudioCodes Gateway Profile

Only three changes need to be made to the AudioCodes Gateway Profile to enable SAS:

First, navigate to "Advanced Parameters" (and "Show Advanced Settings") then the "Stand-Alone Survivability" section at the bottom, and check the "Enable SAS" setting. (Note that the "SAS Local SIP UDP port" is set to 5080, which you used in Step #1.)

Second, navigate to "PSTN to IP Call Routing" and verify the "Failover Mode Routing". You may need to configure destination number manipulation in order to route incoming calls to a registered phone. For example if incoming calls are sent to "6135551234" but the set is registered as "1234" you will need to configure the "Destination Number Manipulation = 613555, *, 6, 0" in order to match the incoming numbers and strip off the leading 6 digits.

Third, navigate to "IP to PSTN Call Routing" and verify the "Destination Number Manipulation". Recall from Step #2 that our sipXecs Dial Plan passes all dialed digits to the gateway. So, you must now configure the gateway to drop the digits that you don't want sent to the PSTN. This way the PSTN calls will work in Normal and Emergency mode, because in both cases it's the gateway dropping the digits.

Refer to the AudioCodes documentation for a full descripion of Destination Number Manipulation. In the example below the AudioCodes Gateway is connected to an analog line where the sipXecs phone can dial "51111" or "63951111" will reach a PSTN phone.

sipXecs Dial Plan:
:built-in Emergency rule: Emergency number "51111", (Optional) PSTN prefix "9".
:Custom "9-ThenAnyNumberOfDigits" rule: Dialed Number Prefix "9" and "Any number of digits", Resulting Call Dial blank and append "Entire Dialed Number"

AudioCodes IP -> Tel number manipulation:

- For each entry the Source Prefix is "*", and the Strip From Right is 0.
- To cover the Emergency rule without the optional PSTN prefix, no manipulation is required.
- To cover the 9-ThenAnyNumberOfDigits rule, and the Emergency rule with the optional PSTN prefix:

- Destination Prefix: 9
- Stripped Digits Number: 1
- Prefix (Suffix) to Add: blank

The resulting manipulations will be:

- Manipulation 1 = "9, *, 1, 0"
- Manipulation 2 = ""
- Manipulation 3 = ""
- Manipulation 4 = ""

Then in both Normal and Emergency mode, composing any of the following dial strings will ring the PSTN phone:

- 51111 (Emergency rule)
- 951111 (Emergency rule, with optional PSTN prefix)
- 963951111 (9-ThenAnyNumberOfDigits rule)

As usual you will need to ensure the Primary DNS setting (under "Network") and the PSTN Lines are both configured.

The profile of course also needs to be re-generated and loaded into the AudioCodes Gateway.