SMART TELECOM

Centrex Prerequisite

The purpose of this document is to specify the operation of VoIP (Voice over IP) services and to provide the necessary prerequisites for the use of our services on routers not managed by Smart Telecom.

On a **link provided by Smart Telecom**, VoIP flows do not pass through the mass of Internet flows. They are routed automatically, via our collection routers at the heart of the network, to our telephone servers. QoS (Quality of Service) rules are applied to the links to ensure regularity of operation.

On a **link NOT provided by Smart Telecom**, VoIP flows pass through the mass of Internet flows. Consequently, the VoIP Smart Telecom service, despite the configuration of routers and switches, can suffer from quality disturbances caused in part by the processing latencies of the flows.

To ensure the proper functioning of telephone stations (or licenses) on a **equipment NOT supplied by Smart Telecom** (Router, Firewall, L3 Switch), you must administer the settings as follows:

- Deactivate the SIP Application Layer Gateway, commonly known as **SIP ALG** (or **NAT Helpers**).

  The SIP ALG function is used to bypass the configuration of static NAT rules on a router. Its implementation varies from router manufacturer to manufacturer, which sometimes makes it difficult to identify. In general, it is necessary to disable SIP ALG and configure port mapping one by one.

- Deactivate **Load Balancing**

  All functions allowing the simultaneous use of several connection links to “speed up” Internet browsing should be avoided. This concerns SD-WAN services.

- On the **Firewall** or the firewall functions of the router used for VoIP services:

  Allow the following **ports**:

  - RTP from 1024 to 65535 (UDP)
  - SIP 5060 → 5062 and 5070 → 5072 (UDP and TCP)
    5063 et 5073 (TLS)
    5074, 5075 (Webex), 5090 (Doko-phone)
  - LDAP 389 (Phonebook access)
  - NTP 123
  - HTTP 80 (Initial provisioning)
  - HTTPS 443
  - XMPP 1081 and 5222
  - SOCKS 52644 and 52645

  For the following **IP ranges**:

  - 217.195.31.128/26
  - 178.255.160.0/24
  - 37.97.64.0/23
  - 81.93.7.0/24
  - 217.74.100.0/23

  FQDN to authorize for **Doko-phone**:

  ws-acro.sewan.fr
  exports.cloudsoftphone.com
  primary.exports.cloudsoftphone.com
  providers.cloudsoftphone.com

- DNS : Data : 178.255.160.92 & 178.255.160.94 VoIP : 178.255.160.91 & 178.255.160.93

In the case of a **shared link** (a single link that includes VOICE and DATA), it may be necessary to set up QoS rules in order to have support for "communication quality" type incidents. The average consumption of a phone equipment is 60kbps. It depends on the audio quality: normal (Codec G.729 = 45kbps), superior audio (codec G.711 = 110kbps) or high definition (Codec G.722 = 110kbps).

For any questions relating to the installation of our VoIP telephony services, we invite you to contact our technical service or send your requests to sav@smart-telecom.fr.