Technical Bulletin	OXO Connect

TC2503 ed.01 Release 3.0 and above

OXO Connect - SIP Advanced Parameters

This document provides a sub-set of system configuration parameters for SIP Trunking and SIP Phones. These parameters were former system flags (noteworthy addresses).

Revision History

Edition 01: September 4, 2018 creation of the document

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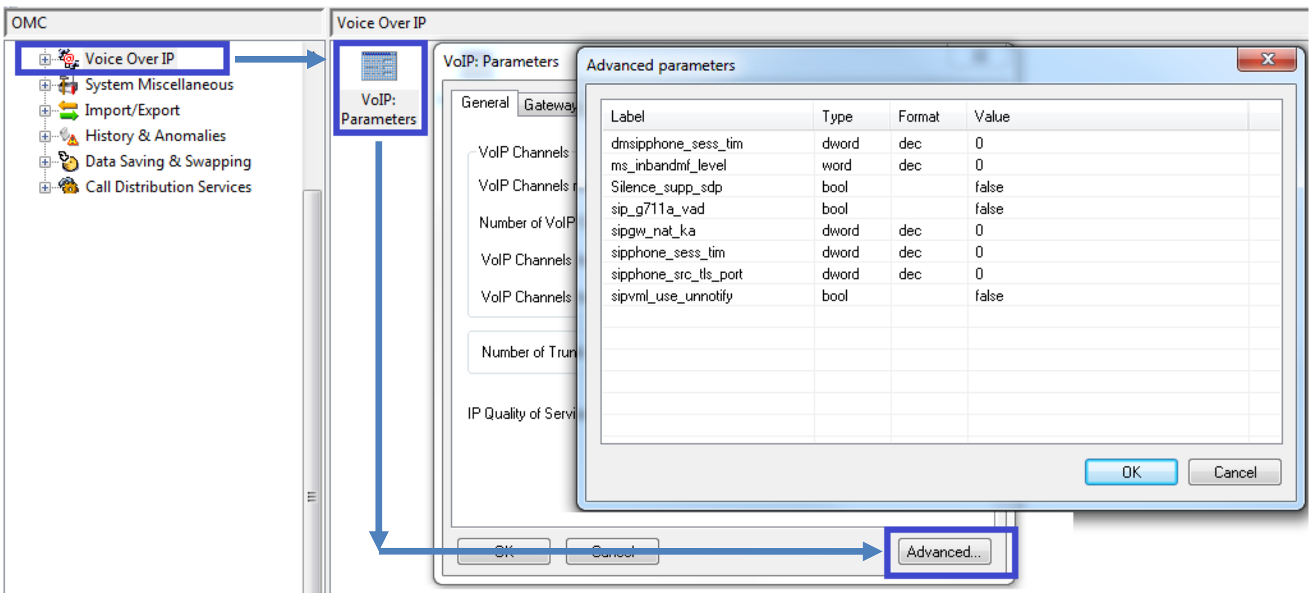
1 Introduction

Since OXO Connect R3.0 (and OXO Connect Evolution) and multiple SIP trunk feature, OMC was redesigned. The former specific SIP Trunking system noteworthy addresses are now displayed as configuration parameters.
This document resumes their signification and value.

2 Voice over IP - Advanced parameters

2.1 Location



2.2 Parameters

These parameters are unique for the whole system. They apply to all SIP Gateways or SIP phones connected to the system.

Warning **A warm reset is needed to take into account these parameters modification!**

2.2.1 dmsipphone_sess_tim

Use	Default	Range
SIP Companion only	0	[0-10800]

Description

The session timer is a parameter specified by RFC4028. It consists in periodic refreshes of SIP sessions to determine if the session is still active. The refresh request, which can be a Re-INVITE or an UPDATE, is performed at 50% of the time specified in this variable. If there is no refresh request received before the end of the timer, the session is considered as terminated and the call is released.

The unit is in second. This timer is the same than “sipphone_sess_tim” but specific to SIP_Companion devices. These devices operate over Wi-Fi. In that configuration, it’s important to monitor the data link and release the call once the user is out of Wi-Fi coverage.

Value

- 0: Use the minimum default value (90s)
- >0: Use this value in seconds as session timer expiration.

2.2.2 ms_inbandmf_level

Use	Default	Range
SIP Trunk and Phone	0	0; [10-40]

Description

Configure the signal magnitude threshold for inband DTMF detection.

Value

- 0: Use the minimum default value (20)
- 10 to 40: Available range.

Warning **Modify only if requested by ALE support !**

2.2.3 Silence_supp_sdp

Use	Default	Range
SIP Phone	False	True; False

Description

Enable/disable "silenceSupp:off" attribute in SDP.

Value

- False: "silenceSupp:off" attribute added in SDP based on VAD
- True: "silenceSupp:off" attribute not added in SDP

2.2.4 sip_g711a_vad

Use	Default	Range
SIP Trunk and Phone	False	True; False

Description

Disable/enable the silence suppression for G711A codec in both Sip Phone and SIP Trunk gateway

Value

- False: Silence Suppression (VAD) value in OMC External Lines->SIP->SIP Gateway->Media, will be used for G711A codec
- True: Silence Suppression (VAD) is disabled for G711a codec only.

2.2.5 sipgw_nat_ka

Use	Default	Range
SIP Trunk	120	[0-65535]

Description

When OXO is behind a router (NAT/Firewall) it must maintain the NAT connection open. This noteworthy address is the duration of the NAT connection is the router. If DNS SRV is enabled, OXO will send OPTION messages at 75% of this delay to avoid the NAT connection to be removed.

Value

- 0: no NAT Keep Alive.
- >0: NAT Keep Alive is enabled for DNS SRV rules, and specifies the NAT connection duration in Seconds.

2.2.6 sipphone_sess_tim

Use	Default	Range
SIP Phone	0	[0-10800]

Description

The session timer is a parameter specified by RFC4028. It consists in periodic refreshes of SIP sessions to determine if the session is still active. The refresh request, which can be a Re-INVITE or an UPDATE, is performed at 50% of the time specified in this variable. If there is no refresh request received before the end of the timer, the session is considered as terminated and the call is released. The unit is in second.

Too high value for the timer is not useful. The refresh requests would not have time to be performed since a call rarely lasts more than 3 hours. The value is only applicable for SIP phones management.

Value

- 0: Use the minimum default value (90s)
- >0: Use this value in seconds as session timer expiration.

2.2.7 sipphone_src_tls_port

Use	Default	Range
SIP Phone	0	[0-65535]

Description

Offers the ability to force the OXO's source port for SIP phone TLS signaling.

Value

- 0: Use the minimum default port (5061 for ie)
- >0: Use this value as source port for SIP Phone TLS signaling. Beware to choose a free port.

Warning **Currently not used. Reserved for future release !**

2.2.8 sipvml_use_unnotify

Use	Default	Range
SIP Phone	False	True; False

Description

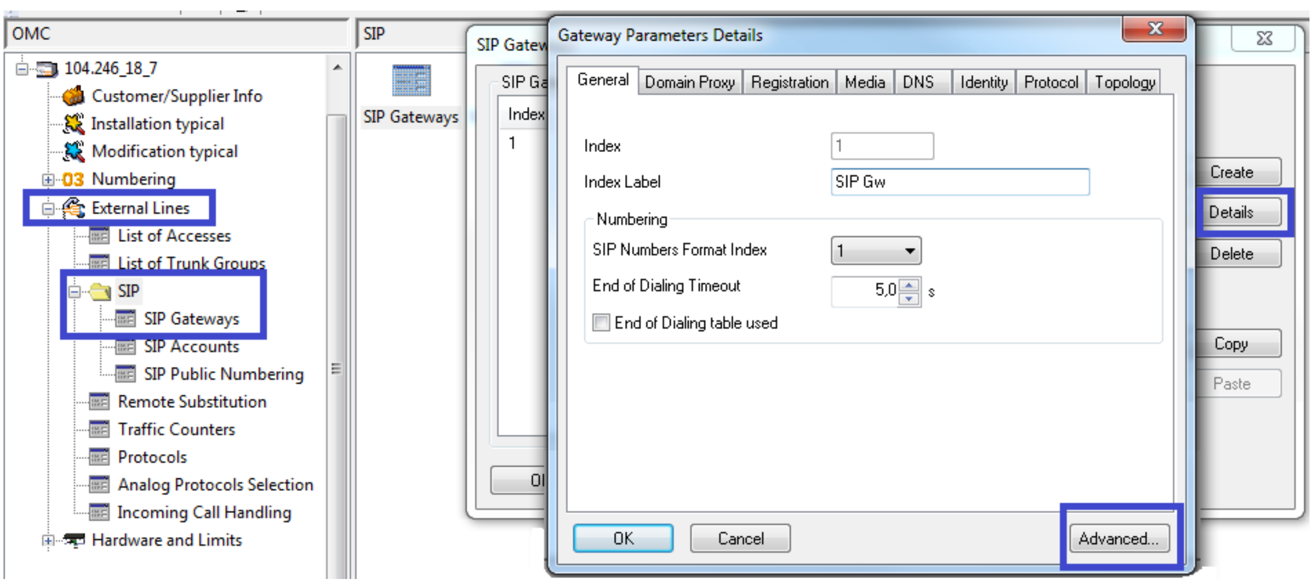
This parameter specifies if OXO uses unsolicited SIP NOTIFY request for the message-summary event for the SIP voicemail application.

Value

- False: OXO uses SIP SUBSCRIBE to create a dialog for the NOTIFY requests from the SIP voicemail for the message-summary events.
- True: OXO do not use SIP SUBSCRIBE and accept unsolicited NOTIFY requests from the SPI voicemail for the message-summary events.
  signaling.

3 SIP Gateway - Advanced parameters

3.1 Location



3.2 Parameters

These parameters are unique for each SIP Gateway.

Note

NO warm reset is needed after these parameters modification!

3.2.1 Auth_optimize

Use	Default	Range
SIP Trunk	False	True; False

Description

Enable/disable the support of Optimization of Authentication.
After a first authenticated request, the system will send a new request with the cached authentication parameters (nonce caching).

Value

- False: Optimization of Authentication is enabled
- True: Optimization of Authentication is disabled

3.2.2 ExtNuFoVoip

Use	Default	Range
SIP Trunk	22	00;02;11;22;33

Description

Type of CLIP build for a SIP trunk user.

Value

- 00: only DDI part of the number
- 02: zone prefix + Installation No. + DDI No.
- 11: subscriber = zone prefix + Installation No. + DDI No.
- 22: zone prefix + Installation No. + DDI No.
- 33: national code/zone prefix + Installation No. + DDI No.

3.2.3 FaxPasCd

Use	Default	Range
SIP Trunk	01ff	01ff;10ff;0fff;1fff

Function

Preferences of codecs when OXO sends a Re-Invite with Fax G711 Offer on detecting fax.

Value

- 01ff: 8(G711alaw), 0(G711ulaw)
- 10ff: 0(G711ulaw), 8(G711alaw)
- 0fff: 8(G711alaw)
- 1fff: 0(G711ulaw)

3.2.4 inhibit_t38

Use	Default	Range
SIP Trunk	0	[0-2]

Function

Fax are transmitted with T.38 protocol by default.

Value

- 00: T.38 FAX protocol : (use the OMC/Gateway parameter to set up the FAX transmission mode)
- 01: No T.38 FAX protocol (use the OMC/Gateway parameter to set up the FAX transmission mode)
- 02: No T.38 FAX detection when G711 mode is configured.

3.2.5 initial_reg_username

Use	Default	Range
SIP Trunk	False	True;False

Function

The initial REGISTER request sent by the UE doesn't include an authentication field. When receiving a reject response with a nonce, the UE send a second REGISTER with an authentication field including the private user identity (username).

With this parameter, the private user identity can be sent in the first REGISTER authentication field.

This is to comply with the 3 GPP TS 24.229 specification.

Value

- False: no authentication field with username in the first REGISTER sent by the UE
- True: the first REGISTER sent by the UE includes an authentication field including the username

3.2.6 INVwSDPtrk

Use	Default	Range
SIP Trunk	False	True;False

Function

In case of joining, OXO will send an INVITE with SDP on the second call leg.

Value

- False: INVITE without SDP sent
- True: INVITE with SDP sent

3.2.7 MultAnsReinv

Use	Default	Range
SIP Trunk	True	True;False

Function

Re-Invite management in case of multiple codec answers

Value

- True: Re-invite sent on multiple-codec answer
- False: No Re-invite sent on multiple-codec answer

3.2.8 multiple_option_req

Use	Default	Range
SIP Trunk	False	True;False

Function

Send SIP OPTION request to all the IP Address present in the DNS Cache.

This parameter will work only if following parameters are configured in OXO:
- "DNSSRV" in Gateway Parameter is enabled.
- "sipgw_nat_ka" is configured with a non-zero value. SIP Option request is then triggered based on the keep alive time interval value configured in "sipgw_nat_ka".

Value

- False: SIP Option request will not be sent to all DNS Cache IP address
- True: OXO will send SIP Option request to all the IP address present in the DNS Cache

3.2.9 MYICcaller

Use	Default	Range
SIP Trunk	00	00;01;02

Function

INVITE & Re-INVITE's FROM value in case of MyIC/OTCV call on first call leg.

Value

- 00: Final called party's ID (Default value)
- 01: From in INVITE contains MyIC's calling number
- 02: From in INVITE & ReINVITE contains MyIC's calling number

3.2.10 no_rport

Use	Default	Range
SIP Trunk	0	[0-2]

Description

This parameter permits to disable the SIP stack rport.

Value

- 0: rport is activated in UDP
- 1: rport is disabled in UDP and TCP
- 2: rport is activated in TCP

3.2.11 PrefCodec

Use	Default	Range
SIP Trunk	0000	0000;[0002-0005]

Description

Force the use of a Codec in case of multiple answers.

Note **Is relevant only if “PrefFramin” is enabled.**

Value

- 0000: disabled
- 0002: G723
- 0003: G729
- 0004: G711 A Law
- 0005: G711 µ Law

3.2.12 PrefFraming

Use	Default	Range
SIP Trunk	0	[0-120]

Description

Force the use of a Framing in case of multiple answers

Note **Is relevant only if “PrefCodec” is enabled.**

Value

- 0: disabled
- 10: 10ms
- 20: 20ms
- Etc…

3.2.13 PrimaryGW

Use	Default	Range
SIP Trunk	0	[0-200]

Description

Offers the possibility to link a secondary gateway with a primary gateway.
The second one is used as a backup when the primary gateway is failing (after keepalive mechanism).
Multiple secondary gateways are allowed per primary. The selection order of these gateways is the one displayed in OMC 'SIP Gateways' list.

Value

- 0: disabled (no primary gateway)
- >0: index of the primary gateway (OMC ordered list)

3.2.14 rfc4916_off

Use	Default	Range
SIP Trunk	0	[0-2]

Description

If UPDATE method allowed on OXO, then UPDATE without SDP messages are sent by OXO when connected identities change.
(RFC4916 configuration).

Value

- 0: sending of these UPDATE messages is enabled.
- 1: sending of these UPDATE messages is disabled
- 2: OXO does not support "from-change" and UPDATE message is disabled

3.2.15 SimulIpAlt

Use	Default	Range
SIP Trunk	False	True;False

Description

Enable/disable the external ring back tone simulation. (in case of receiving a 183 without SDP).

Value

- False: OXO doesn't generate a ring back tone
- True: OXO generates a local ring back tone.

3.2.16 sip_capa

Use	Default	Range
SIP Trunk	False	True;False

Description

Display the sip capabilities in the contact header of the register request and also in the 200 OK response to an OPTION request.

Value

- False: Presence of a sip_capabilities in the contact header of the register request.
- True: No Presence of a sip_capabilities in the contact header of the register request

3.2.17 SIPdtmfInB

Use	Default	Range
SIP Trunk	False	True;False

Description

Manage the inbound « inband" DTMF signaling on a public SIP trunk.
Only valid when "out of band" DTMF is configured for the trunk.

Value

- False: Inband DTMF ignored
- True: Inband DTMF taken into account

3.2.18 sipgw_fax_offer

Use	Default	Range
SIP Trunk	False	True;False

Description

Allow the user to negotiate the re-invite offer as Media Fax or Media Audio.

Value

- False: Call treated as a FAX when OXO receives Audio as First MEDIA and Fax as next MEDIA.
  Call treated as a FAX when OXO receives FAX as the first MEDIA with/without Audio as next MEDIA.
- True: Call treated as an AUDIO when OXO receives Audio as First MEDIA and Fax as next MEDIA.
  Call treated as a FAX when OXO receives FAX as the first MEDIA with/without Audio as next MEDIA

3.2.19 sipgw_namedisp

Use	Default	Range

SIP Trunk	False	True;False

Description

If the SIP header of an incoming INVITE contains *From: "name" <anonymous@anonymous.invalid>*.

Value

- False: "name" is not displayed.
- True: "name" is displayed.

3.2.20 sipgw_prefid

Use	Default	Range
SIP Trunk	0	0;16

Description

Building the "Display info" of the From header during an outgoing INVITE request.

Value

- 0: The Display Info of the From contains the sets usual name
- 16: The Display Info contains the users DDI number if configured, else it contains the AOR

3.2.21 sipgw_priv_lvl

Use	Default	Range
SIP Trunk	False	True;False

Description

Set the OXO privacy policy when CLIR is used. If identity presentation of user 1234 is restricted, the From field of an outgoing INVITE from the OXO will be:

Value

- False: From: sip:anonymous@anonymous.invalid
- True: From: sip:1234@LocalDomain

3.2.22 sipgw_reg_trigger_01 to sipgw_reg_trigger_10

Use	Default	Range
SIP Trunk	0	0;[400-699]

Description

This parameter is an array of ten elements: Each element is a failure response code (4xx or 5xx or 6xx). This list of cause allows the triggering of registration when an INVITE requests fails with an error response code, in the context of a proxy failover using DNS.

Value

- 0: Empty element
- xxx: SIP failure response code (4xx, 5xx, 6xx) coded in decimal**...**

3.2.23 sipgw_regid

Use	Default	Range
SIP Trunk	0	0;8;16

Description
Use of the Address of Record (AOR) in Invite:

Value

- 0: No AOR. DID number of user is sent along with the INVITE message
- 8 or 16: AOR in all headers configured in OMC for all outgoing INVITE request.

3.2.24 sipgw_rem_maxptime

Use	Default	Range
SIP Trunk	0	[0-2]

Description

Option to remove maxptime attribute present in the SDP of all INVITE offers from OXO in SIP Trunk Gateway. This maxptime attribute will be removed only when codec framing is forced in ARS and not when “Default” is selected.

Value

- 0: maxptime attribute will be present in all INVITE SDP offers
- 1: maxptime attribute will be absent in initial INVITE SDP offer.
- 2: maxptime attribute will be absent in all INVITE SDP offers

3.2.25 sipgw_Req_URI_route_call

Use	Default	Range
SIP Trunk	False	True;False

Description

Option to process different To and Req_URI header for display info.
Hence, Call routing will be based on Req_URI header only.

Value

- False: Call routing is based on the OMC parameter “Routing on ‘To’ Header” and the display will be a “forwarding” like if ‘Req_URI’ and ‘To’ are different.
- True: Call routing is based on the Req_URI (whatever is the value of OMC parameter “Routing on ‘To’ Header” and the display will ignore the header difference, but will consider the ‘Diversion’ or ‘History-info’ header.

3.2.26 sipgw_to_ruri

Use	Default	Range
SIP Trunk	False	True;False

Description

Option for building the “To” field for outgoing SIP calls.

Value

- False: INVITE 1234@LocalDomain & To: sip:5678@LocalDomain
- True: INVITE 1234@LocalDomain To: sip:1234@LocalDomain

3.2.27 sipgw_voip_caun

Use	Default	Range
SIP Trunk	False	True;False

Description

Call server sends destination out of order as release cause.

Value

- False: 502 Bad gateway
- True: 404 Not found

3.2.28 SIPInDspNm

Use	Default	Range
SIP Trunk	0	0;1;3

Description

Display or not the CNIP (Name) received from the SIP network

Value

- 0: CNIP not displayed for both PUBLIC/PRIVATE SIP calls.
- 1: CNIP displayed for both PUBLIC/PRIVATE SIP calls
- 3: CNIP displayed for PRIVATE and not displayed for the PUBLIC calls.

3.2.29 SIPOgDspNm

Use	Default	Range
SIP Trunk	0	0;1;3

Description

Send or not the CNIP (Name) onto the SIP network.

Value

- 0: CNIP not sent for both PUBLIC/PRIVATE SIP Outgoing calls
- 1: CNIP sent for both PUBLIC/PRIVATE SIP Outgoing calls
- 3: CNIP sent for PRIVATE calls and not sent for PUBLIC calls

3.2.30 Special_char_truncation

Use	Default	Range
SIP Trunk	False	True;False

Description

Enable special char truncation in outgoing called number.

Value

- False: NO truncation of special character and further digit(s)/char(s) in called numbers.

- True: HASH(#) char and further digit(s)/char(s) are truncated in called numbers

3.2.31 SuprAlerTone

Use	Default	Range
SIP Trunk	False	True;False

Description

Offers the ability to play or not the Alert tone when we receive the 180 ringing request.

Value

- False: NO truncation of special character and further digit(s)/char(s) in called numbers.
- True: HASH(#) char and further digit(s)/char(s) are truncated in called numbers

3.2.32 T4_jit

Use	Default	Range
SIP Trunk	0	[0-65535]

Description

Offers the possibility to tune the depth of the fax T4 jitter buffer.

Value

- 0: Default T4 jitter buffer depth applies, i.e 240ms
- >0: Depth of the T4 jitter buffer depth in milliseconds

3.2.33 trigger_alert

Use	Default	Range
SIP Trunk	False	True;False

Description

Offers the possibility to send 180 Ringing and 183 Session Progress, when 180 Ringing or 183 Session Progress with SDP is received, with the delay (~1s) after 100 Trying is received in transit case.

Value

- False: In transit case, if 180 Ringing or 183 Session Progress is received with SDP with a delay (~1s) after receiving 100 Trying, OXO sends Session Progress (183) with SDP to another call leg.
- True: transit case, if 180 Ringing or 183 Session Progress is received with SDP with a delay (~1s) after receiving 100 Trying, OXO sends Alert message (180 Ringing), which is followed by Session Progress (183) with SDP to another call leg.

3.2.34 USalterfrom

Use	Default	Range
SIP Trunk	False	True;False

Description

For US country incoming calls to subscriber, in case of transfer, the second call-leg FROM field contains the first call-leg calling party information.

Value

- False: Normal behavior
- True: Second call-leg FROM contains calling party of first call-leg

3.2.35 userlvlpri

Use	Default	Range
SIP Trunk	False	True;False

Description

Allows to enable user level privacy for identity secrecy calls.

Value

- False: User level Privacy is enabled
- True: User level Privacy is disabled

3.2.36 V21_jit

Use	Default	Range
SIP Trunk	0	[0-65535]

Description

Offers the possibility to tune the depth of the fax V21 jitter buffet.

Value

- 0: Default V21 jitter buffer depth applies, i.e 240ms
- >0: Depth of the V21 jitter buffer depth in milliseconds

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