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Technical Communication	TC 1765	Ed 01	Date: 21/01/2013
Product: Alcatel OmniPCX Office			**Nb. of pages: 12**
Subject: OmniPCX Office SIP Trunking R8.X/R9.0 Noteworthy Addresses			

Introduction

- The various Label Addresses in the Alcatel OmniPCX Office can be modified via OMC.
- Only modify the addresses given here, taking care with the spelling of labels (some labels are similar).
- When modifying time delays, never enter the value FFFFh.
- New values entered are saved after a warm reset (unless otherwise stated), but are lost after a cold reset.
- After making changes to some addresses you may need to perform a warm reset of the system in order to enable the new values (it's stated if a warm reset is required).
- The values of the labels described in this document are given for information purposes. Some values can be different according to the country or according to the OmniPCX Office release.

Noteworthy addresses via OMC

a) In "System Miscellaneous" – "Memory Read/Write" choose the type of address to be modified ("Other Labels" / "Timer Labels" or "Debug Labels"), select the label of the address to modify and click "Details".

b) In the upper part of the window, change the value of the desired byte and click "Modify" then "Write".

History

Edition	Modification
01	Document creation

1. "Other Labels" description

Label	Function description	No of bytes	Default value
AlCodLst	Modify the default codec list priority for outgoing calls : **00**=G729a-30 / G723.1-30 / G711-30 **01 :** G723.1-30 / G729a-30 / G711-30 **02 :** G723.1-30 / G711-30 / G729a-30 **03 :** G711-30 / G723.1-30 / G729a-30 **04 :** G711-30 1G729a-30 2G723.1-30 **05 :** G729a-30 / G711-30 / G723.1-30 **06 :** G711-30 / G729a-30 **07 :** G729a-30 / G711-30 **08 :** G711-30 / G723.1-30 **09 :** G723.1-30 / 1G711-30 **0A :** G723.1-30 / G729a-30 **0B :** G729a-30 / G723.1-30 **16 :** G711-20 / G729a-20	**1**	**00**
DtmfDynPL	Value of the Dynamic Paylod for DTMF-RFC4733 **XX** : Hexadecimal value to convert in decimal **6A** : 106 decimal value **78** : 120 decimal value	**1**	**6A**
ExtNuFVoi	Type of CLIP build for a SIP trunk user. **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.	**1**	**22**
FaxPasCd	Preferences of codecs when OXO sends a Re-Invite with Fax G711 Offer on detecting fax. **01FF** : 8(G711alaw), 0(G711ulaw) **10FF** : 0(G711ulaw), 8(G711alaw) **0FFF** : 8(G711alaw) **1FFF** : 0(G711ulaw)	**2**	**01 FF**
PrefCodec	Force the use of a Codec in case of multiple answers **00 00** : disabled **02 00** : G723 **03 00** : G729 **04 00** : G711 A Law **05 00** : G711 μ Law ***Note : Is relevant only if "PrefFramin" is enabled***	**2**	**00 00**
PrefFramin	Force the use of a Framing in case of multiple answers **00** = disabled **0A** = 10 decimal = 10 ms **14** = 20 decimal = 20ms etc… ***Note : Is relevant only if "PrefCodec" is enabled***	**1**	**00**

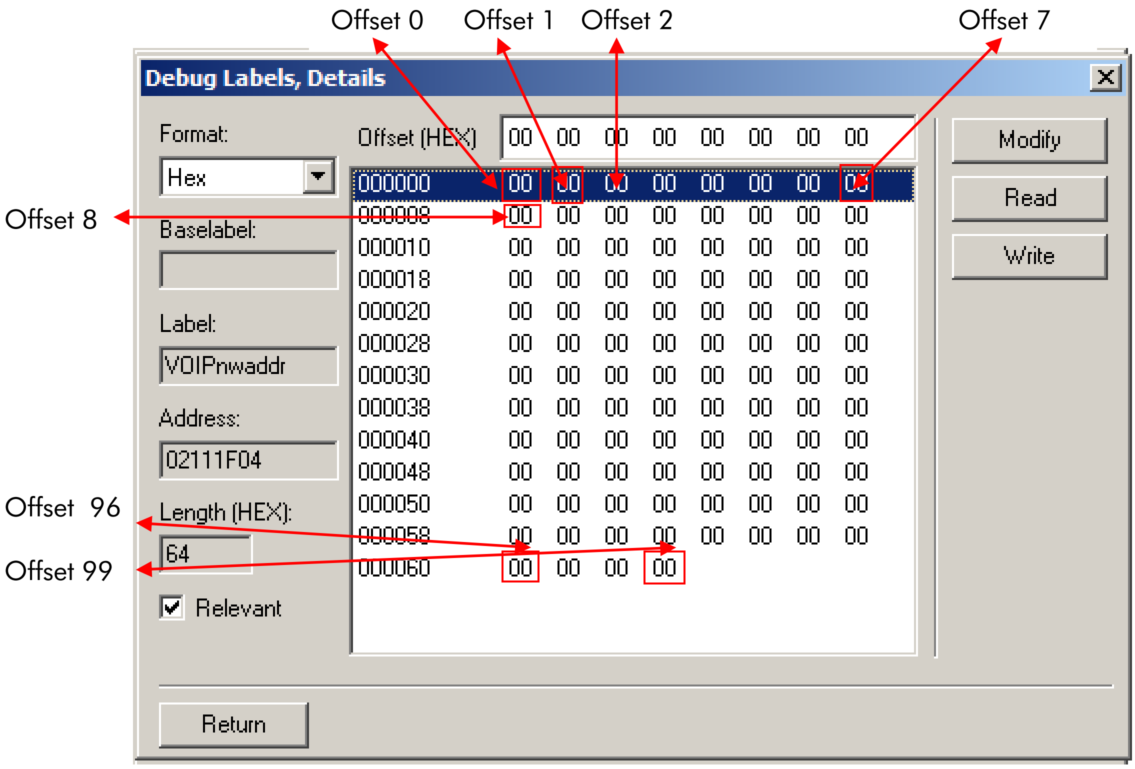
Label	Function description	No of bytes	Default value
VipPuNua	Controls how the incoming called numbers are managed (private or public) on the SIP trunk **00** : Called number based on the SIP Prefixes (both types –Private & Public) **01** : Called number based on the SIP trunk type (only one –Private or Public)	**1**	**00**
SimulIpAlt	Enable/disable the external ring back tone simulation. (in case of 183 without RTP). **00**: OXO generates **nothing** **01**: OXO **generates** a **local** ring back tone.	**1**	**00**
SIPInDspNm	Display or not the CNIP (Name) received from the SIP network **00**: CNIP not displayed for both PUBLIC/PRIVATE SIP calls. **01**: CNIP displayed for both PUBLIC/PRIVATE SIP calls **03**: CNIP displayed for PRIVATE and not displayed for the PUBLIC calls.	**1**	**01**
SIPOgDspNm	Send or not the CNIP (Name) onto the SIP network . Value **00**: CNIP not sent for both PUBLIC/PRIVATE SIP Outgoing calls Value **01**: CNIP sent for both PUBLIC/PRIVATE SIP Outgoing calls Value **03**: CNIP sent for PRIVATE calls and not sent for PUBLIC calls	**1**	**01**

2. "Debug Labels" description

Label	Function description	No of bytes	Default value
MultAnsReinv	Re-Invite management in case of multiple codec answers **01** : Re-invite sent on multiple-codec answer **00** : No Re-invite sent on multiple-codec answer	**1**	**01**
SuprAlerTo	Play or not the Alert tone when we receive the 180 ringing request. **01** : Don't play the Alert Tone **00** : Play the Alert Tone	**1**	**00**
VOIPnwaddr	Allows tuning of the SIP protocol on IP Trunks.	**64**	See Appendix 3.1 for details

3. APPENDIX

3.1 VOIPnwaddr



3.1.1 Remote SIP Port

Offset **0 & 1**	On 2 bytes	Default value is **00 00**	**Obsolete since R6.0**

Description

This value sets the remote port number used by SIP trunk. The outgoing Invite will be sent to the defined port value . If the value is **0**, the default SIP will be used (**5060**).

3.1.2 Privacy Level

Offset **2**	On 1 byte	Default value is **00**	/

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 :

- **00** : From: sip:anonymous@anonymous.invalid
- **01** : From: sip:1234@LocalDomain

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3.1.3 To as Req-URI

Offset **3**	On 1 byte	Default value is **00**	/

Description

This flag is used for building the "To" field for outgoing SIP calls. By default, the "To" field is the same as "Request URI" **except for private forward**. In this case, the "To" field is the diverting number (the destination of the first call). To fix the "To" field to be identical as "Req-UR"I, the value must be set to 1.

- **00 :** INVITE 1234@LocalDomain & To: sip:5678@LocalDomain
- **01** : INVITE 1234@LocalDomain To: sip:1234@LocalDomain

Note: if called is public, the To field will be always the same as the Req URI.

3.1.4 Session Timer

Offset **4 & 5**	On 2 bytes	Default value is **00 00**	/

Description

The session timer is the delay parameter specified by RFC4028. In the context of each call, a keep alive (session refresh) is performed at 50% of the period specified by this variable : it consists in a Re-Invite or an Update. If no refresh is performed / successful at the end of this timer, the call is released. The unit is the minute.

- **0000** : Default timer, which is 720 minutes (**43200** sec = 12 hours),
- **FFFF** : D**isable** Session Timer.
- **xxxx** : Any other value specifies the period in minutes unit.

3.1.5 Don't use DNS SRV unreachable proxy list

Offset **6**	On 1 byte	Default value is **00**	/

Description

DNS SRV makes use of a quarantine list to memorize the unreachable proxies : it optimizes overflow by not trying known unreachable proxies.

- **00** : Use unreachable proxy list.
- **01** : Don't use unreachable proxy list.

3.1.6 T.38 FAX inhibition

Offset **7**	On 1 byte	Default value is **00**	**Obsolete / Use gateway parameter !**

Description

FAX are transmitted with T.38 protocol by default

- **00** : No changes (T.38 FAX protocol is allowed)
- **01** : No T.38 FAX protocol (but in-band FAX)

3.1.7 NAT Keep Alive for DNS SRV

Offset **8 & 9**	On 2 bytes	Default value is **00 00**	/

Description

In context of DNS-SRV and OXO placed behind a NAT/Firewall OXO will send OPTION messages at 75% of this delay to avoid the NAT connection to be removed.

- **00 00** : no NAT Keep Alive.
- **02 58** : Timer is 0258 Hex = 600 decimal = 600 seconds for ie
- **xx xx** : NAT Keep Alive is enabled for DNS SRV rules, and specifies the NAT connection duration

3.1.8 SIP Trunk Signaling Source Port

Offset **10 & 11**	On 2 bytes	Default value is **00 00**	**Obsolete since R710 / port fixed to 5060**

Description

Force the OXO's source port for SIP signalling. This parameter is associated to both UDP and TCP transport.

- **00 00** : The source port is SIP stack dependent, it uses either a dynamically allocated source port or a static 5060 source Port (Sip stack version dependant).
- **13 C4** : Hex value is 5060 Decimal value = So port is 5060 for ie.
- **xx xx** : Use this port value as source for SIP signalling

3.1.9 Rport activation

Offset **12**	On 1 byte	Default value is **00**	/

Description

This parameter permits to disable the SIP stack rport.

- **00**: rport is activated
- **01**: rport is disabled

3.1.10 Unavailable Voip Cause

Offset **13**	On 1 byte	Default value is **00**	/

Description

If Call server sends destination out of order as release cause :

- **00**: SIP gw sends 404 Not found to the network.
- **01**: SIP gw sends 502 Bad gateway.

3.1.11 Name display with CLIR

Offset **14**	On 1 byte	Default value is **00**	**/**

Description

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

- **00**: "name" is not displayed.
- **01**: "name" is displayed.

3.1.12 Registration identifier

Offset **15**	On 1 byte	Default value is **00**	**Value 8 introduced since R820 only**

Description

Use of the AOR in Invite, if :

- **00**: No AOR. DID number of user is sent along with the INVITE message
- **01**: Use the registration AOR in *P-asserted id* for outgoing INVITE.
- **02**: Use the registration *Contact* in outgoing INVITE's *Contact* header.
- **03**: Use Registration AOR in *P-asserted id* and registration *Contact* in INVITE's Contact (cumulate 01&02).
- **04**: Use the registration AOR in *P-preferred id* for outgoing INVITE.
- **08**: Use the registration AOR in P-asserted id and contact header for all INVITE request sent (Re-Invite).

3.1.13 P-Prefered id

Offset **16**	On 1 byte	Default value is **00**	/

Description

Use of P_prefered & P_asserted headers

- **00**: Outgoing calls: no *P-Pi* / Incoming Calls: Process *P-Pi* only *if P-AI* is absent.
- **01**: Outgoing Calls: add a *P-Pi* with user identity (DID) / Incoming calls: Process *P-Pi* only if *P-Ai* is absent.
- **02**: Outgoing Calls: no *P-Pi* / Incoming calls: Process *P-Pi* in priority.
- **03**: Outgoing Calls: add a *P-Pi* with user identity (DID) / Incoming calls: Process *P-Pi* in priority.
- **04**: Outgoing Calls: no *P-Pi*. No *P-Ai &* Incoming calls: Process *From* header in priority.
- **05**: Outgoing Calls: add a *P-Pi* with user identity (DID). No *P-Ai &* Incoming calls: Process *From* header in priority.

3.1.14 Support of the Update Method

Offset **17**	On 1 byte	Default value is **00**	/

Description

Enable/disable the support of the Update method.

- **00**: OXO is **allowed** to send or receive "UPDATE" method (default value)
- **01**: OXO is **not allowed** to send or receive "UPDATE" method

3.1.15 TCP switching

Offset **18**	On 1 byte	Default value is **00**	/

Description

Enable/disable the UDP to TCP switching.

- **00**: OXO is **allowed** to switch from UDP to TCP.
- **01**: OXO is **not allowed** to switch from UDP to TCP

Note : Value 01 sets the flag "Do Not Fragment" to false (in IP Layer)

3.1.16 Send No-Signal

Offset **19**	On 1 byte	Default value is **00**	**Value 02 introduced in R900**

Description

Send no-signal message for outgoing fax call.

- **00**: OXO does not send no-signal message
- **01**: OXO send no-signal messages for outgoing fax call
- **02**: OXO sends CNG messages for outgoing fax call

3.1.17 History-info

Offset **20**	On 1 byte	Default value is **00**	/

Description

Enable/disable the support of History-Info header (RFC4244) at transmit and receive.

- **00**: History-Info processing is enabled
- **01**: History-Info processing is disabled

3.1.18 Optimization of authentication

Offset **21**	On 1 byte	Default value is **00**	/

Description

Enable/disable the support of Optimization of Authentication .

- **00**: Optimization of Authentication is enabled
- **01**: Optimization of Authentication is disabled

3.1.19 DNS-A resolution

Offset **22**	On 1 byte	Default value is **00**	/

Description

Possibility for OXO to enable the Force DNS A resolution.

- **01**: Force OXO to make DNS A resolution instead of DNS SRV resolution.
- **00**: DNS_SRV is enabled and DNS A resolution is disabled for OXO.However, OXO can make DNS A resolution only when the request for SRV records resulted in "no such record" response.

3.1.20 Registration check

Offset **23**	On 1 byte	Default value is **00**	/

Description

Enable registration check before routing requests (register and non-register) to the outbound proxy.

- **00**: No checking for presence of a valid registration is done before routing requests.
- **01**: Presence of a valid registration is checked before routing non-register requests to the outbound proxy. If a valid registration to the outbound proxy doesn't exist, a REGISTER request is triggered for registration.

3.1.21 V21 Jitter Buffer

Offset **24 & 25**	On 2 bytes	Default value is **00 00**	/

Description

Tune the depth of the fax V21 jitter buffer. The unit is the millisecond.

- **00 00** : Default V21 jitter buffer depth applies, i.e 240ms.
- **02 00** : Hex value is 512 Decimal value = 512ms.
- **xx xx** : Timer value in ms.

3.1.22 T4 Jitter Buffer

Offset **26 & 27**	On 2 bytes	Default value is **00 00**	/

Description

Tune the depth of the fax T4 jitter buffer. The unit is the millisecond.

- **00 00** : Default T4 jitter buffer depth applies, i.e 240ms.
- **02 00** : Hex value is 512 Decimal value = 512ms.
- **xx xx** : Timer value in ms.

3.1.23 Alert Message

Offset **28**	On 1 byte	Default value is **00**	/

Description

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.

- **00**: 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.
- **01**: In 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.

3.1.24 SIP Capabilities

Offset **29**	On 1 byte	Default value is **00**	/

Description

Display the sip capabilities in the contact header of the register request.

- **00**: Presence of a sip_capabilities in the contact header of the register request.
- **01**: No Presence of a sip_capabilities in the contact header of the register request.

3.1.25 Source Port for SIP Phones

Offset **30 & 31**	On 2 bytes	Default value is **00 00**	**Introduced in R810**

Description

Force the OXO's source port for SIP Phone signalling in UDP and TCP.

- **00 00** : The source port is 5059.
- **13 C1** : Hex value is 5057 Decimal value = So port is 5057 for ie.
- **xx xx** : Use this port value as source for SIP signalling

3.1.26 Reserved

Offset **32 & 33**	On 2 bytes	Default value is **00 00**	**Reserved**

Description

Reserved

3.1.27 T38 CED

Offset **34**	On 1 byte	Default value is **00**	/

Description

Send simulated T38 ced message to the network.

- **00**: Simulated T38 ced message will not be sent to the network
- **01**: Simulated T38 ced message will be sent to the network

3.1.28 IP Authentication

Offset **35**	On 1 byte	Default value is **00**	**Introduced in R810**

Description

Enable authentication of incoming call based on IP address for DNS enabled ARS lines.

- **00**: Source IP address is not checked for incoming calls associated to DNS ARS lines.
- **01**: Source IP address is checked for incoming calls associated to DNS ARS lines.

3.1.29 Session Timer for SIP Phones

Offset **36 & 37**	On 2 bytes	Default value is **00 00**	**Introduced in R820**

Description

The session timer is the delay parameter specified by RFC4028. In the context of each call, a keep alive (session refresh) is performed at 50% of the period specified by this variable : it consists in a Re-Invite or an Update. If no refresh is performed / successful at the end of this timer, the call is released. The unit is the minute.

- **00 00** : Default timer, which is 1800 seconds (30mn),
- **2A 30** : Maximum value (3 Hours)
- **xx xx** : Any other value specifies the period in seconds unit.

3.1.30 Registration Triggering causes

Offset **38 to 57**	On 20 bytes	Default is **00 00** **00 00** **00 00** **00 00** **00 00** **00 00** **00 00** **00 00** **00 00** **00 00**	**Introduced in R820**

Description

This parameter is an array of ten elements (of 2 bytes) : 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.

- **00 00** : Empty element
- **01 93** : Means 403 Decimal, is 403 Forbidden.
- **xx xx** : SIP failure response code (4xx, 5xx, 6xx) coded in hexadecimal..

3.1.31 P-early-media Activation

Offset **58**	On 1 byte	Default value is **00**	**Introduced in R900**

Description

This parameter offers the ability to enable the control of early media flow through p-early-media

- **00** : P-Early-Media is disabled, no control of early media flow.
- **01** : P-Early-Media is enabled, control of early media flow.

3.1.32 G711A Silence Suppression

Offset **59**	On 1 byte	Default value is **00**	**Introduced in R900**

Description

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

- **00** : Silence Suppression (VAD) value configured in OMC->Voice Over IP->VoIP: Parameters will be used for G711A codec.
- **01** : Silence Suppression (VAD) is disabled for G711A Codec alone.

3.1.33 FAX ECM

Offset **60**	On 1 byte	Default value is **00**	**Introduced in R900**

Description

disable the FAX ECM mode on SIP or H.323 trunking.

- **00** : Fax ECM is enabled.
- **01** : Fax ECM is disabled.

End of the document