Generic Gateway
Interoperability Tests
Alcatel OmniPCX Enterprise 6.0

Version history:

<table>
<thead>
<tr>
<th>Version</th>
<th>Date/author</th>
<th>Changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.1</td>
<td>18.10.2004/Klaus Darilion</td>
<td>initial release</td>
</tr>
<tr>
<td>0.2</td>
<td>21.12.2004/Klaus Darilion</td>
<td>remove IP addresses</td>
</tr>
</tbody>
</table>
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Date
2004-11-17 – 2004-11-18

Tests performed by
Klaus Darilion, enum.at
Brigitte Steinkellner, Alcatel

Product
Alcatel OmniPCX Enterprise 6.0

Software Version
Alcatel OmniPCX Enterprise 6.0
Release: DELIVERY f1.602
Patch identification: 3
Dynamic patch identification: i
Country: au
Cpu: c80s1
Linux Mandrake for eMediate release 7.2 (Odyssey) for i486
Kernel 2.4.1-ll-dhs3 on an i486 / ttyS0
rack small with 1 Media-Gateway, 1 Call-Server, 1 MIX-Board
1 digital Alcatel reflex set advanced (4035T)
1 analog set
1 attendant set 4035T (operator)
1 advanced e-reflex 4035-IP phone
TSCIP version 02.20.0
hardware version 3AK 23047 ABDA01
boot version 2.00.03

Configuration
Configuration concerning SIP:
RTP direct = YES
GF diversion on joining = NO
DPNSS prefix created
Trunk Groups
   Node Number (reserved) : 105
   Trunk Group ID : 510
   Trunk Group Type + T2
   Trunk Group Name : SIP5
   remote network: 3
   number Compatible With : -1
   Shared Trunk Group + False
   Private Trunk Group + False
   Q931 Signal variant + ABC-F
   Number Of Digits To Send : 0
   Channel selection type + Quantified
   Auto.DTMF dialing on outgoing call + NO
   T2 Specification + SIP
SIP network
   network number: 3
   protocol type: QSIG-GF
   associated ext SIP gateway: -1
SIP gateway
   subnetworknumber: 3
   trunk group: 510
   IP Address: Y.Y.Y.25
SIP user
   Node Number (reserved) : 105
   Directory Number : 509
   Directory name : PINGTEL
   Directory First Name : ---------------------
   Location Node : 5
   Shelf Address : 255
   Board Address : 255
   Equipment Address : 255
   Set Type + Extern Station
   Entity Number : 1
   Set Function + Default
   URL UserName : 509
   URL Domain : Y.Y.Y.25

Tested Protocols

SIP

Gateway Version

Cisco Internetwork Operating System Software
IOS (tm) 5300 Software (C5300-IS-M), Version 12.3(10), RELEASE SOFTWARE (fc3)
Copyright (c) 1986-2004 by cisco Systems, Inc.
Compiled Tue 17-Aug-04 00:54 by kellythw
Image text-base: 0x60008AEC, data-base: 0x615EC000

ROM: System Bootstrap, Version 12.0(2)XD1, EARLY DEPLOYMENT RELEASE SOFTWARE (fc1)
BOOTLDR: 5300 Software (C5300-BOOT-M), Version 12.0(2)XD1, EARLY DEPLOYMENT RELEASE SOFTWARE (fc1)
gggw1 uptime is 7 weeks, 22 hours, 40 minutes
System returned to ROM by reload at 13:00:20 GMT Thu Sep 30 2004
System restarted at 13:01:10 GMT Thu Sep 30 2004
System image file is "flash:c5300-is-mz.123-10.bin"
cisco AS5300 (R4K) processor (revision A.32) with 131072K/16384K bytes of memory.  
Processor board ID 14051896
R4700 CPU at 150MHz, Implementation 33, Rev 1.0, 512KB L2 Cache
Channelized E1, Version 1.0.
Bridging software.
X.25 software, Version 3.0.0.
SuperLAT software (copyright 1990 by Meridian Technology Corp).
Primary Rate ISDN software, Version 1.1.
Backplane revision 2
Manufacture Cookie Info:
EEPROM Type 0x0001, EEPROM Version 0x01, Board ID 0x30,
Board Hardware Version 3.2, Item Number 73-2414-06,
Board Revision C0, Serial Number 14051896,
1 Ethernet/IEEE 802.3 interface(s)
1 FastEthernet/IEEE 802.3 interface(s)
35 Serial network interface(s)
4 Channelized E1/PRI port(s)
30 DSP(s), 60 Voice resource(s)
128K bytes of non-volatile configuration memory.
32768K bytes of processor board System flash (Read/Write)
8192K bytes of processor board Boot flash (Read/Write)

Configuration register is 0x2102

**Gateway Configuration**

Current configuration : 6413 bytes

| Gateway Configuration |

| Current configuration : 6413 bytes |
| ! |
| ! Last configuration change at 14:31:33 GMT Thu Nov 18 2004 |
| ! NVRAM config last updated at 14:45:33 GMT Thu Nov 11 2004 |
| ! version 12.3 |
| no service pad |
| service timestamps debug datetime msec localtime show-timezone |
| service timestamps log datetime msec localtime show-timezone |
| service password-encryption |
| ! |
| hostname gggw1 |
| ! |
| boot-start-marker |
| boot system flash c5300-is-mz.123-10.bin |
| boot-end-marker |
| ! |
| logging buffered 100000 debugging |
| enable secret 5 $XXXXXXXXXXXXXXXXXXXXXX/$ |
| enable password 7 XXXXXXXXXXXXXXX |
| ! |
username mrs password 7 XXXXXXXXXXXXX
!
resource-pool disable
clock timezone GMT 0
!
 aaa new-model
!
!
aaa authentication login default enable
aaa authentication enable default enable
aaa authorization exec enum group radius
aaa accounting update periodic 1
aaa accounting connection voip start-stop group radius
aaa session-id common
ip subnet-zero
ip domain name labs.nic.at
ip name-server X.X.X.X
!
!
isdn switch-type primary-net5
!
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 5 gsmfr
 codec preference 10 g711alaw
 codec preference 11 g711ulaw
!
voice class codec 3
 codec preference 1 g711alaw

voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g729br8
 codec preference 3 g711alaw
 codec preference 4 g711ulaw
 codec preference 6 gsmfr
 codec preference 7 g723ar53
 codec preference 8 g723ar63
 codec preference 9 g723r53
 codec preference 10 g723r63
!
!
!
voice class h323 13
 call start slow
 h245 caps mode restricted
!
!
!
!
voice enum-match-table 1
  rule 1 1 /^1\(.*\)/ /\+\1/ e164test.labs.nic.at
  rule 2 2 /^2\(.*\)/ /\+\1/ e164test.labs.nic.at

voice enum-match-table 2
  rule 1 1 /^\(.*\)/ /\+\1/ e164test.labs.nic.at

voice enum-match-table 3
  rule 1 1 /^\(.*\)/ /\+\1/ e164.arpa

voice enum-match-table 4
  rule 1 1 /^\(.*\)/ /\+\1/ enum.telekom.at

voice translation-rule 12
  rule 1 /^1/ //
  rule 2 /^2/ //

voice translation-rule 13
  rule 1 /^\(.*\)/ /00\1/ type international unknown

voice translation-rule 14
  rule 1 /^\(.*\)/ /1\1/

voice translation-profile 12
  translate calling 13
  translate called 12
  translate redirect-called 14

voice translation-profile 13
  translate redirect-called 14

fax interface-type fax-mail
  mta send server X.X.X.232 port 25
  mta send subject Fax message from gateway
  mta send filename incoming_fax date
  mta send mail-from hostname gggw1
  mta send mail-from username $s$

controller E1 0
  clock source line primary
tri-group timeslots 1-31

controller E1 1
  shutdown
clock source line secondary 1

controller E1 2
  shutdown
clock source line secondary 2
controller E1 3
shutdown
clock source line secondary 3
gw-accounting aaa
method voip
acct-template callhistory-detail
!

translation-rule 12
Rule 0 ^143780 43780 international unknown
!

translation-rule 13
Rule 0 ^243780 43780 international unknown
!

translation-rule 22
Rule 0 ^1878102843 878102843 international unknown
!

translation-rule 1
Rule 1 222 431 national international
Rule 2 % 43 national international
!

translation-rule 2
Rule 0 % 00 international unknown
!

translation-rule 14
Rule 0 ^10 0 international unknown
Rule 1 ^11 1 international unknown
Rule 2 ^12 2 international unknown
Rule 3 ^13 3 international unknown
Rule 4 ^14 4 international unknown
Rule 5 ^15 5 international unknown
Rule 6 ^16 6 international unknown
Rule 7 ^17 7 international unknown
Rule 8 ^18 8 international unknown
Rule 9 ^19 9 international unknown
!

translation-rule 15
Rule 0 ^20 0 international unknown
Rule 1 ^21 1 international unknown
Rule 2 ^22 2 international unknown
Rule 3 ^23 3 international unknown
Rule 4 ^24 4 international unknown
Rule 5 ^25 5 international unknown
Rule 6 ^26 6 international unknown
Rule 7 ^27 7 international unknown
Rule 8 ^28 8 international unknown
Interface Ethernet0
  no ip address
  no ip route-cache
  shutdown

Interface Serial0
  no ip address
  no ip route-cache
  shutdown
clockrate 2015232
  no fair-queue

Interface Serial1
  no ip address
  no ip route-cache
  shutdown
clockrate 2015232
  no fair-queue

Interface Serial2
  no ip address
  no ip route-cache
  shutdown
clockrate 2015232
  no fair-queue

Interface Serial3
  no ip address
  no ip route-cache
  shutdown
clockrate 2015232
  no fair-queue

Interface Serial0:15
  no ip address
  isdn switch-type primary-net5
isdn timer t309 2000
  isdn overlap-receiving T302 5000
  isdn incoming-voice modem
  no cdp enable

Interface FastEthernet0
  ip address X.X.X.231 255.255.255.192
  no ip route-cache
  load-interval 30
duplex auto
  speed auto

ip default-gateway X.X.X.254
ip classless
ip route 0.0.0.0 0.0.0.0 X.X.X.254
no ip http server
!
ip rtcp report interval 10000
!
logging facility local5
!
snmp-server enable traps tty
!
radius-server host X.X.X.234 auth-port 1645 acct-port 1646 key 7 0876141E0009
radius-server vsa send accounting
!
call application voice enumfax tftp://X.X.X.232/enumfax.tcl
!
voice-port 0:D
  translate calling 1
  input gain 2
  echo-cancel coverage 32
  echo-cancel suppressor
  timeouts interdigit 2
!
!
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
  description Default incoming dial-peer from PSTN
  application enumfax
  incoming called-number 43T
  direct-inward-dial
!
!
dial-peer voice 12 voip
  translation-profile outgoing 12
  preference 1
  destination-pattern 1T
  voice-class codec 2
  session protocol sipv2
  session target enum:2
!
!
dial-peer voice 13 voip
  translation-profile outgoing 12
  preference 2
  destination-pattern 2T
  voice-class codec 3
  voice-class h323 13
  session target enum:2
  no vad
!
!
dial-peer voice 14 mmoip
  preference 3
  application fax_on_vfc_onramp_app out-bound
destination-pattern 43T
information-type fax
session target mailto:$d$@faxrelay

gateway
timer receive-rtcp 5
!
sip-ua
  no remote-party-id
  set sip-status 401 pstn-cause 127
  set sip-status 407 pstn-cause 127
  set sip-status 410 pstn-cause 22
  set sip-status 415 pstn-cause 127
  set sip-status 480 pstn-cause 19
  set sip-status 503 pstn-cause 127
  set sip-status 580 pstn-cause 127
retry invite 2
retry response 2
retry bye 2
retry cancel 2
!
!
line con 0
  password 7 XXXXXXX
line aux 0
  password 7 XXXXXXXX
line vty 0 4
  exec-timeout 0 0
  timeout login response 300
  password 7 XXXXXXX
!
scheduler interval 1000
ntp clock-period 17179120
ntp server X.X.X.2
end

ENUM Entries
; direct calling
9.0.5.0.5.3.4 NAPTR 100 10 "u" "E2U+sip" "!^.*$!sip:509@Y.Y.Y.25!" .
1.0.5.0.5.3.4 NAPTR 100 10 "u" "E2U+sip" "!^.*$!sip:501@Y.Y.Y.25!" .

; wildcard calling
*.1.5.3.4 NAPTR 100 10 "u" "E2U+sip" "!^\+4351(.*)$!sip:\1@Y.Y.Y.25!" .

Test Scenarios
• Basic call (PSTN ==> Generic Gateway ==> Alcatel OmniPCX ==> various Phones) in the following scenarios:

<table>
<thead>
<tr>
<th>No</th>
<th>Scenario</th>
<th>Test Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>A calls B, B does not answer, A hangs up</td>
<td>- B should start ringing</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- B should stop ringing after A hangs up</td>
</tr>
<tr>
<td>No</td>
<td>Scenario</td>
<td>Test Purpose</td>
</tr>
<tr>
<td>----</td>
<td>---------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| 2  | A calls B, B answers, A hangs up                  | - B should start ringing  
- establish full-duplex voice path  
- B should receive “Call Disconnected” indication (e. g. Busy Tone or visual indication) when A hangs up |
| 3  | A calls B, B answers, B hangs up                  | - B should start ringing  
- establish full-duplex voice path  
- A should receive “Call Disconnected” indication (e. g. Busy Tone or visual indication) when B hangs up |
| 4  | A calls B, B answers, C calls B                   | - Busy Tone indication at C                                                                                                                                                                               |
| 5  | A calls B, B redirects call to C without answering the call | - Test the redirect feature of the gateway                                                                                                                                                                |

- **Codec Tests**  
All supported codecs were tested for interoperability by human listening tests.

- **DTMF**  
DTMF was tested in direction PSTN $\Rightarrow$ SIP by human listening for incoming DTMF tones.

**Note:** Call transfer of established calls from one extension to another extension is not supported by the gateway and therefore was not tested.

**Test Setup**

The “big picture”:

![Diagram of test setup](image-url)
The actual test setup:

The tests were performed by calling the phone number 01 79453 4351 *abc* where *abc* is the particular extension of the phone set.

**Additional Devices**

Pingtel SIP Phone:
- Pingtel XPRESSA Phone Model PX-1
- Core Apps: 2.1.11.24, AUG 20 2003 19:34:46
- Kernel: 2.1.11.24, AUG 20 2003 19:28:16
- serial: 00d01e005670
- revision: 960-001-907 C 4002

Cisco SIP Phone:
- Cisco CP-7960G
- Application Load ID: POS3-06-2-00
- Boot Load ID: PC030301
- Dsp Load ID: PS03AT38
- serial: INMO7421JJQ
- 68-1679-07 Rev. D0

**Tests**

The tests consist of two tests: the signaling test and the codec test.
Signaling Test

<table>
<thead>
<tr>
<th>Phone (B) Scenario</th>
<th>Pingtel ext. 509</th>
<th>Cisco 7960 ext. 505</th>
<th>Alcatel IP ext. 501</th>
<th>Alcatel ext. 500</th>
<th>Analog Phone ext. 510</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>2</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>3</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>4</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>5</td>
<td>OK</td>
<td>not supported</td>
<td>OK</td>
<td>OK</td>
<td>not tested</td>
</tr>
</tbody>
</table>

Comments:
1. The phones do not show the calling number id (which is in the From: header of the INVITE message from the gateway) but instead shows the trunk id of the SIP network configured in the OmniPCX.
   Comments from Alcatel concerning the display name presentation on Alcatel phone sets:
   - In the code, we found that:
     - if there is a name in MGR, we send it. --> in our case no SIP set registration inside the PBX
     - else, if there is a display name (configured directly in the SIP set), we send it.
     - else, if the trunk has a name, we send it, --> this was the case in the interoperability test.
     - else, send "SIP".

2. If B hangs up, it takes 3-5 seconds until the BYE message will be sent to the gateway
3. The behavior of “how to deal with concurrent incoming calls” is configurable in all devices. Typical options are e. g. “send busy” or “call waiting indication.

Codec Test

Syntax:
- OK Codec supported by B and tested succesfully
- n/s Codec not supported by B, therefore no test performed
- X Codec supported by B but not interoperable with the gateway

<table>
<thead>
<tr>
<th>Phone (B) Codec</th>
<th>Pingtel ext. 509</th>
<th>Cisco 7960 ext. 505</th>
<th>Alcatel IP ext. 501</th>
<th>Alcatel ext. 500</th>
<th>Analog Phone ext. 510</th>
</tr>
</thead>
<tbody>
<tr>
<td>g729r8</td>
<td>n/s</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>g729br8</td>
<td>n/s</td>
<td>OK</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g711alaw</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>g711ulaw</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>gsmefr</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>gsmfr</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g723ar53</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g723ar63</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g723r53</td>
<td>n/s</td>
<td>n/s</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>g723r63</td>
<td>n/s</td>
<td>n/s</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
<tr>
<td>Codec</td>
<td>Pingtel ext. 509</td>
<td>Cisco 7960 ext. 505</td>
<td>Alcatel IP ext. 501</td>
<td>Alcatel ext. 500</td>
<td>Analog Phone ext. 510</td>
</tr>
<tr>
<td>--------</td>
<td>-----------------</td>
<td>---------------------</td>
<td>---------------------</td>
<td>-----------------</td>
<td>----------------------</td>
</tr>
<tr>
<td>g726r16</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g726r24</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g726r32</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
<tr>
<td>g728</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
<td>n/s</td>
</tr>
</tbody>
</table>

Comments:
- OmniPCX supports G.729 and G.723, but not concurrently. Only one of them may be used. This is a configuration option.
- The Pingtel SIP phone responds with “400 Bad Request” to INVITE messages with non-supported codecs instead of using “488 Not Acceptable Here”.

**DTMF Test**

<table>
<thead>
<tr>
<th>Phone (B) DTMF</th>
<th>Pingtel ext. 509</th>
<th>Cisco 7960 ext. 505</th>
<th>Alcatel IP ext. 501</th>
<th>Alcatel ext. 500</th>
<th>Analog Phone ext. 510</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
<td>OK</td>
</tr>
</tbody>
</table>

Comments:
- The DTMF recognition was only tested by “human ears”.

**General Comments**
- Calls to an extension which is configured in the OmniPCX but without attached phone set: OmniPCX responses with “100 Trying”, but then there is no other action. After a timeout the gateway sends a CANCEL and terminates the call.
- Calls to a non existing extension (not configured): OmniPCX responded with “410 Gone” or with “480 Temporarily Unavailable”. Apparently this is a configuration option. Maybe in some circumstances it should respond with “404 Not Found”.
- If the OmniPCX receives a UDP SIP message which exceeds a certain size, the proxy forwards the message with TCP. The limit for the transport protocol switch-over is somewhere between 1062 and 1153 Bytes UDP payload. This is not necessary as the maximum UDP payload size in Ethernet is 1472 Bytes. Although fragmented UDP packets can be used (might raise problems when firewalls and NAT devices are involved). This automatic switchover prevents the usage of phone set which only supports UDP (e.g. Cisco 7960G).

Comments from Alcatel:

(See RFC3261 chapter 18 and 18.1).

*All SIP elements MUST implement UDP and TCP. SIP elements MAY implement other protocols. Making TCP mandatory for the UA is a substantial change from RFC 2543. It has arisen out of the need to handle larger messages, which MUST use TCP, as discussed below. Thus, even if an element never sends large messages, it may receive one and needs to be able to handle them.*

*If a request is within 200 bytes of the path MTU, or if it is larger than 1300 bytes and the path MTU is unknown, the request MUST be sent using an RFC 2914 [43] congestion controlled transport protocol, such as TCP. If this causes a change in the*
transport protocol from the one indicated in the top Via, the value in the top Via MUST be changed. This prevents fragmentation of messages over UDP and provides congestion control for larger messages. However, implementations MUST be able to handle messages up to the maximum datagram packet size. For UDP, this size is 65,535 bytes, including IP and UDP headers.

The 200 byte "buffer" between the message size and the MTU accommodates the fact that the response in SIP can be larger than the request. This happens due to the addition of Record-Route header field values to the responses to INVITE, for example. With the extra buffer, the response can be about 170 bytes larger than the request, and still not be fragmented on IPv4 (about 30 bytes is consumed by IP/UDP, assuming no IPSec). 1300 is chosen when path MTU is not known, based on the assumption of a 1500 byte Ethernet MTU.