

# Generic Gateway Interoperability Tests

Alcatel OmniPCX Enterprise 6.0



## Version history:

<i>Version</i>	<i>Date/author</i>	<i>Changes</i>
0.1	18.10.2004/Klaus Darilion	initial release
0.2	21.12.2004/Klaus Darilion	remove IP addresses

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## **Date**

2004-11-17 – 2004-11-18

## **Tests performed by**

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## **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,  
PLD/ISP Version <unset>, Manufacture Date 23-Jun-1999.  
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  
!  
! 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 \$XXXXXXXXXXXXXXXXXXXXX/  
enable password 7 XXXXXXXXXXXXXXXX  
!

```
username mrs password 7 XXXXXXXXXXXXXXXX
!  
!  
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 $$  
!  
!  
controller E1 0  
clock source line primary  
pri-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
```

Rule 9 ^29 9 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 XXXXXX
line aux 0
password 7 XXXXXXXX
line vty 0 4
exec-timeout 0 0
timeout login response 300
password 7 XXXXXX
!
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:

<i>No</i>	<i>Scenario</i>	<i>Test Purpose</i>
1	A calls B, B does not answer, A hangs up	- B should start ringing - B should stop ringing after A hangs up

<i>No</i>	<i>Scenario</i>	<i>Test Purpose</i>
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.

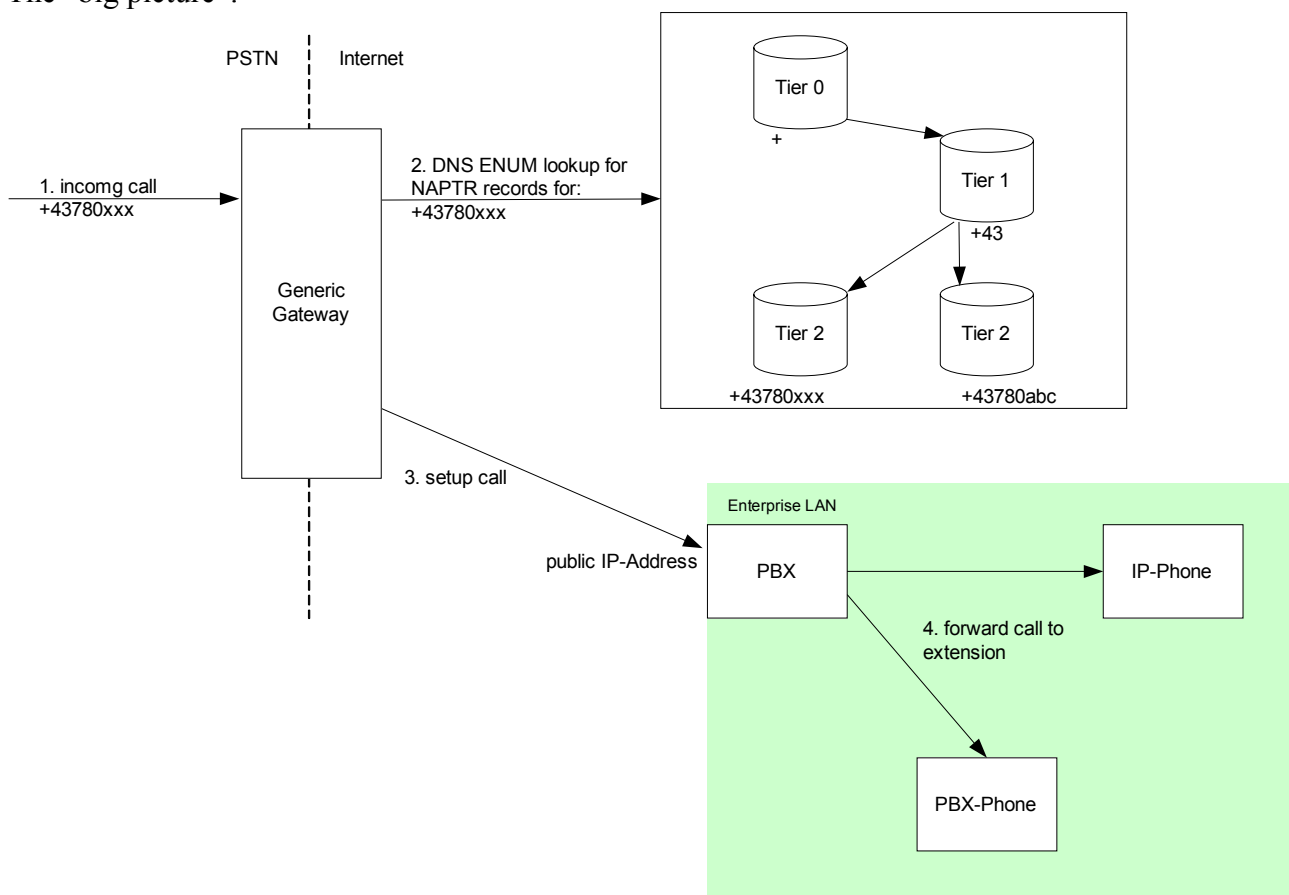
- **DTMF**

DTMF was tested in direction PSTN ==> SIP by human listening for incoming DTMF tones.

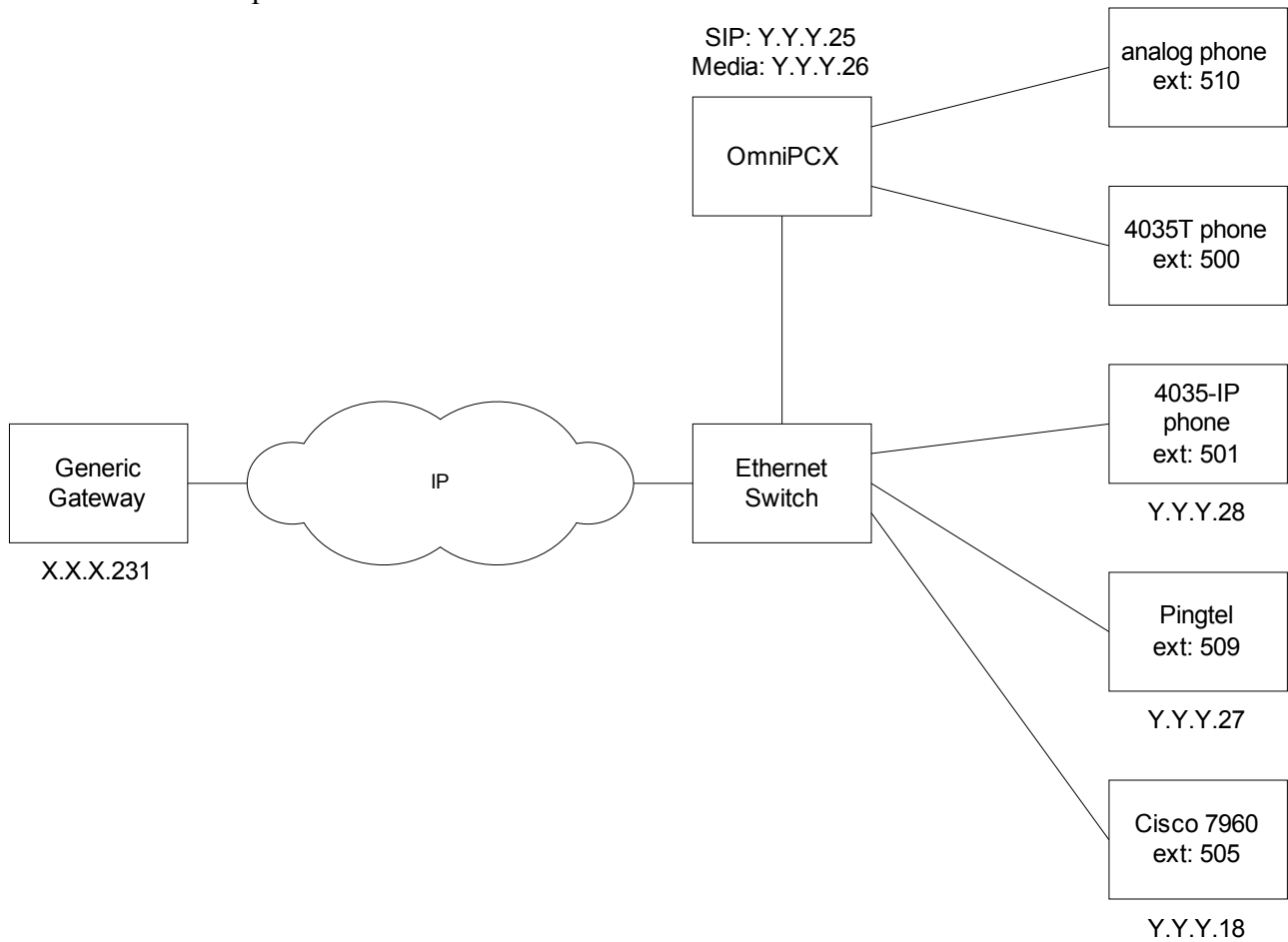
**Note:** Call transfer of established calls from one extension to another extension is not supported by the gateway and therefore was not tested.

## Test Setup

The "big picture":



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

<i>Phone (B) Scenario</i>	<i>Pingtel ext. 509</i>	<i>Cisco 7960 ext. 505</i>	<i>Alcatel IP ext. 501</i>	<i>Alcatel ext. 500</i>	<i>Analog Phone ext. 510</i>
1	OK	OK	OK <sup>1</sup>	OK <sup>1</sup>	OK
2	OK	OK	OK <sup>1</sup>	OK <sup>1</sup>	OK
3	OK	OK	OK <sup>1,2</sup>	OK <sup>1</sup>	OK
4 <sup>3</sup>	OK	OK	OK <sup>1</sup>	OK <sup>1</sup>	OK
5	OK	not supported	OK <sup>1</sup>	OK <sup>1</sup>	not tested

### Comments:

- 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".*

- If B hangs up, it takes 3-5 seconds until the BYE message will be sent to the gateway
- 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 successfully
- n/s Codec not supported by B, therefore no test performed
- X Codec supported by B but not interoperable with the gateway

<i>Phone (B) Codec</i>	<i>Pingtel ext. 509</i>	<i>Cisco 7960 ext. 505</i>	<i>Alcatel IP ext. 501</i>	<i>Alcatel ext. 500</i>	<i>Analog Phone ext. 510</i>
g729r8	n/s	OK	OK	OK	OK
g729br8	n/s	OK	n/s	n/s	n/s
g711alaw	OK	OK	OK	OK	OK
g711ulaw	OK	OK	OK	OK	OK
gsmefr	n/s	n/s	n/s	n/s	n/s
gsmfr	n/s	n/s	n/s	n/s	n/s
g723ar53	n/s	n/s	n/s	n/s	n/s
g723ar63	n/s	n/s	n/s	n/s	n/s
g723r53	n/s	n/s	OK	OK	OK
g723r63	n/s	n/s	OK	OK	OK

<i>Phone (B) Codec</i>	<i>Pingtel ext. 509</i>	<i>Cisco 7960 ext. 505</i>	<i>Alcatel IP ext. 501</i>	<i>Alcatel ext. 500</i>	<i>Analog Phone ext. 510</i>
g726r16	n/s	n/s	n/s	n/s	n/s
g726r24	n/s	n/s	n/s	n/s	n/s
g726r32	n/s	n/s	n/s	n/s	n/s
g728	n/s	n/s	n/s	n/s	n/s

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

<i>Phone (B) DTMF</i>	<i>Pingtel ext. 509</i>	<i>Cisco 7960 ext. 505</i>	<i>Alcatel IP ext. 501</i>	<i>Alcatel ext. 500</i>	<i>Analog Phone ext. 510</i>
DTMF	OK	OK	OK	OK	OK

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.*