Generic Gateway Interoperability Tests

Alcatel OmniPCX Enterprise 6.0

Version history:

Version	*Date/author*	*Changes*
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 attandant 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 $XXXXXXXXXXXXXXXXXXX/
enable password 7 XXXXXXXXXXXXX
!
```

```
username mrs password 7 XXXXXXXXXXXXXX
!
!
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 gatetway
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
 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 XXXXXXX
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:

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

No	Scenario	Test Purpose
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.

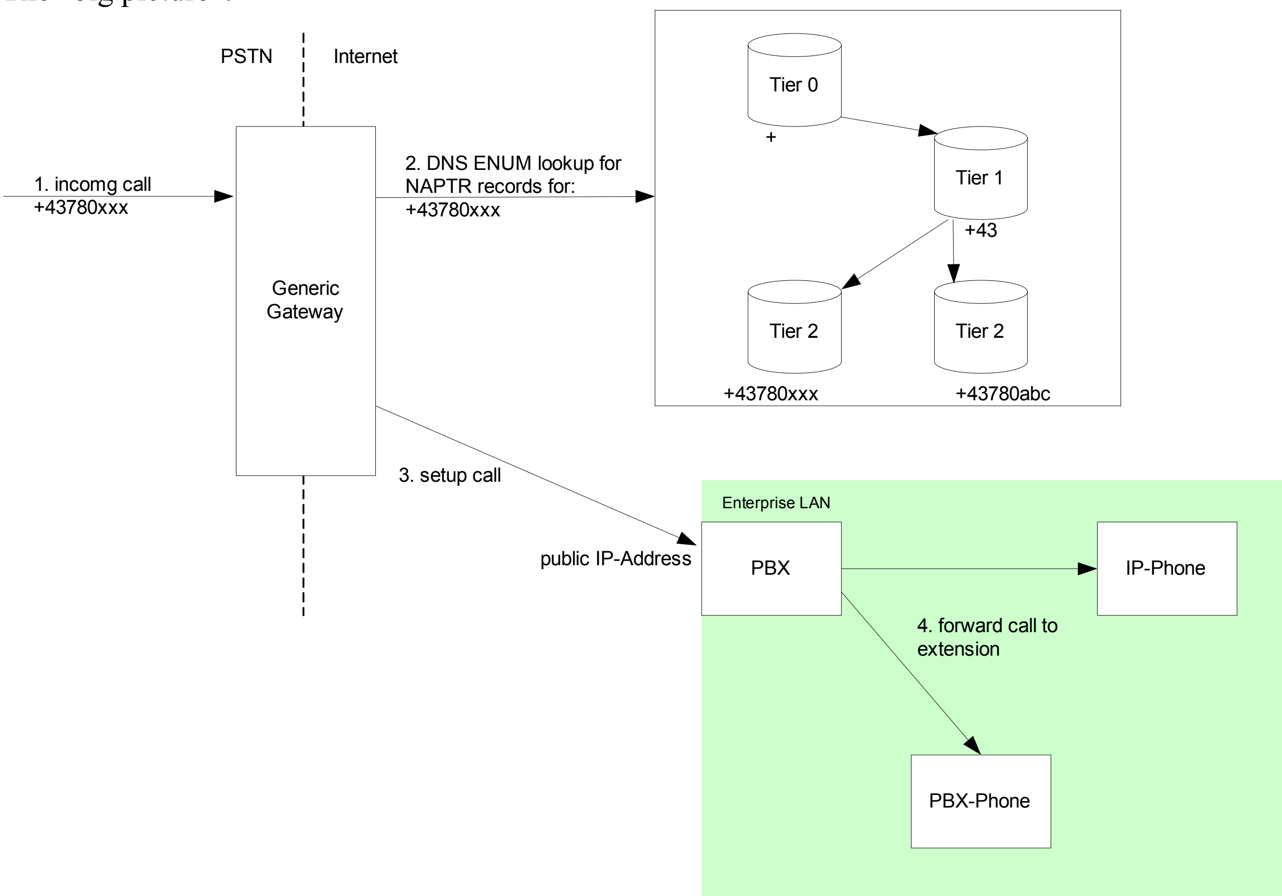
- DTMF

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

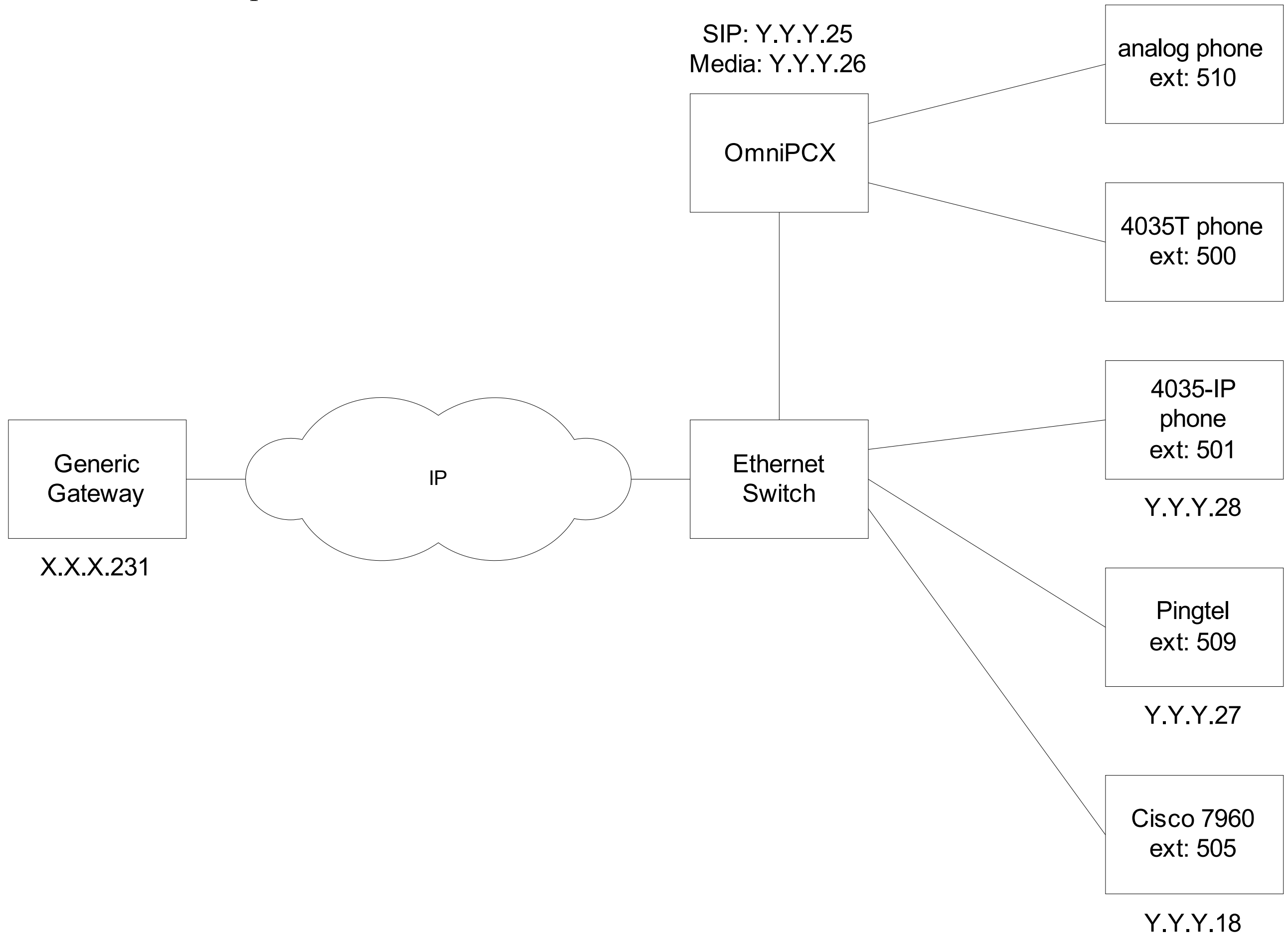
Note: Call transfer of established calls from one extension to another extension is not supported by the gateway and therefore was not tested.

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

Phone (B) *Scenario*	*Pingtel* *ext. 509*	*Cisco 7960* *ext. 505*	*Alcatel IP* *ext. 501*	*Alcatel* *ext. 500*	*Analog Phone* *ext. 510*
1	OK	OK	OK [1]	OK [1]	OK
2	OK	OK	OK [1]	OK [1]	OK
3	OK	OK	OK [1,2]	OK [1]	OK
4 [3]	OK	OK	OK [1]	OK [1]	OK
5	OK	not supported	OK [1]	OK [1]	not tested

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

Phone (B) *Codec*	*Pingtel* *ext. 509*	*Cisco 7960* *ext. 505*	*Alcatel IP* *ext. 501*	*Alcatel* *ext. 500*	*Analog Phone* *ext. 510*
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

Phone (B) *Codec*	*Pingtel ext. 509*	*Cisco 7960 ext. 505*	*Alcatel IP ext. 501*	*Alcatel ext. 500*	*Analog Phone ext. 510*
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

Phone (B) *DTMF*	*Pingtel ext. 509*	*Cisco 7960 ext. 505*	*Alcatel IP ext. 501*	*Alcatel ext. 500*	*Analog Phone ext. 510*
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.