Audiocodes TP-260

Asterisk Interoperability Report
February 2007

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Asterisk Interoperability Reports describe the certification testing performed by Digium on the specified product and Asterisk Business Edition. Each Supported feature of the device and/or service under test is described as well as how the device/service was configured to work with Asterisk during testing.

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Device Summary

Device Type:	Stand-Alone VoIP Media Gateway
Device Model:	TP-260 SIP
Asterisk Version:	Business Edition B.1.2

Device Description

The TP-260 is a stand-alone SIP/T1 media gateway PCI card. Compliant with the SIP protocol, the Digital Media Gateway features 1, 2, 4 or 8 digital T1/E1 ports and connects PSTN lines to Asterisk BE using T1 connectivity.

Functionality Testing Overview

The following test cases were performed in order to confirm testing performed by Audiocodes and reported in Audiocodes' Interoperability Test Plan Document #: LTRT-50703.

- **SIP Registration/De-Registration**
- **IP->PSTN Call Routing**
- **PSTN->IP Call Routing**
- **Codec Compatibility**
- **DTMF Transmission/Reception**

Asterisk Configuration

The following Asterisk configuration files were edited as shown in order to perform the Interoperability testing. First in sip.conf the audiocodes peer is defined, so that succesful registration occurs. The extensions.conf file contains the dial plan logic portion of the test setup.

sip.conf

```
[audiocodes]
context=sip-trunk
dtmfmode=rfc2833
host=dynamic
qualify=yes
nat=never
type=peer

[audiocodes]
context=sip-trunk
dtmfmode=rfc2833
host=dynamic
type=user
```

extensions.conf

```
[default]
;Dial Audiocodes from sip to test IP->PSTN
exten => 1100,1,Dial(sip/${EXTEN}@audiocodes)
exten => 1111,1,Dial(sip/${EXTEN}@audiocodes)
;Dial zap group 1 to test PSTN->IP
exten => 1200,1,Dial(Zap/g1/${EXTEN})
exten => 1212,1,Dial(Zap/g1/${EXTEN})
```

```
[sip-trunk]
;Playback tt-weasels when PSTN->IP successful
exten => 1200,1,Background(demo-congrats)
exten => 1200,n,Hangup
;Read DTMF Test (PSTN->IP)
exten => 1212,1,Read(testvar)
exten => 1212,n,NoOp(${testvar})
exten => 1212,n,Hangup

[t1-trunk]
;Playback tt-monkey when IP->PSTN successful
exten => 1100,1,Background(tt-monkeys)
exten => 1100,n,Hangup
;Read DTMF Test (IP->PSTN)
exten => 1111,1,Read(testvar2)
exten => 1111,n,NoOp(${testvar2})
exten => 1111,n,Hangup
```

Asterisk Configuration continued

The following configures a Digium T1 card to establish a trunk with the TP-260.

zaptel.conf

```
loadzone = us
defaultzone=us

span=1,0,0,esf,b8zs
bchan=1-23
dchan=24
```

zapata.conf

```
[trunkgroups]

[channels]
context=t1-trunk
switchtype=national
signalling=pri_net
usecallerid=yes
hidecallerid=no
callwaiting=yes
usecallingpres=yes
callwaitingcallerid=yes
threewaycalling=yes
transfer=yes
canpark=yes
cancallforward=yes
callreturn=yes
echocancel=yes
echocancelwhenbridged=yes
group=1
immediate=no
channel => 1-23
```

Device Configuration

Device Configuration Overview

Configuration of Audiocodes devices takes place in two locations; within the device configuration (.ini) files, and within the web administration tool.

Audiocodes BOARD.ini Configuration File

Configuration files provide an easy way to save configurations from the web interface and then can be loaded within the same interface. This keeps the web-admin tool at the center of configuration, but does allow for saving and editing clear-text configuration files.

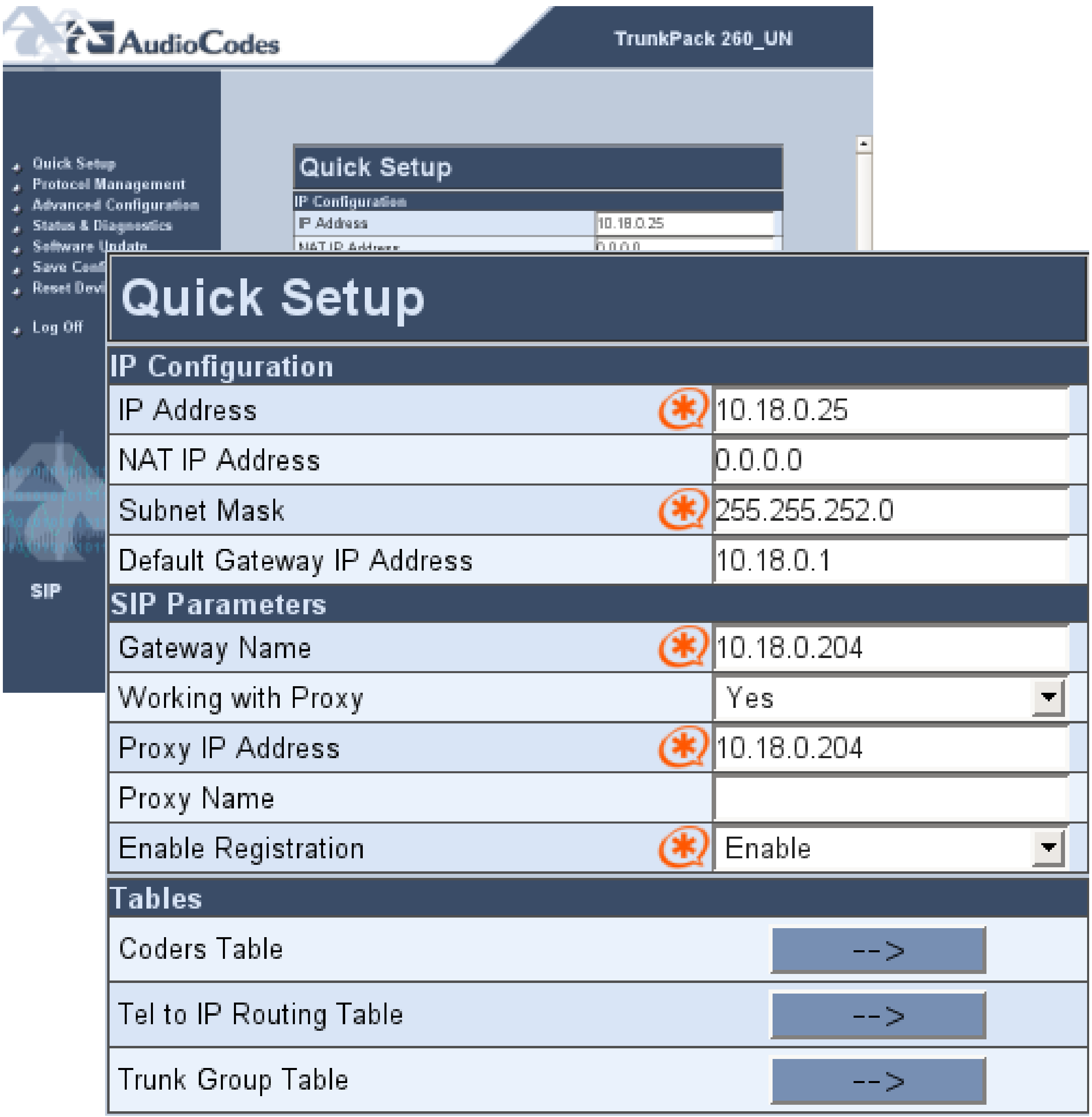
```
;**************
;** Ini File **
;**************

;Board: TrunkPack 260_UN
;Serial Number: 719384
;Slot Number: 1
;Software Version: 4.80A.033
;Board IP Address: 10.18.0.25
;Board Subnet Mask: 255.255.252.0
;Board Default Gateway: 10.18.0.1
;Ram size: 128M   Flash size: 8M
;Num DSPs: 12  Num DSP channels: 31
;Profile: NONE
;Key features:;Max SW Ver: 9.99;Board Type: TrunkPack 260_UN;Control Protoc$
;------------------------------


[SYSTEM Params]

DNSPriServerIP = 10.2.1.2
SyslogServerIP = 10.18.3.199
EnableSyslog = 1
DisableRS232 = 1

[BSP Params]

PCMLawSelect = 3
TDMBusType = 2
TDMBusClockSource = 4
LocalOAMIPAddress = 10.18.0.25
RoutingTableHopsCountColumn = 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, $
```

BOARD.ini (continued)

```
[PSTN Params]

ProtocolType = 10
FramingMethod = C

[SS7 Params]


[Voice Engine Params]

IdlePCMPattern = 85
VoiceVolume = 1
FaxRelayRedundancyDepth = 2
FaxRelayEnhancedRedundancyDepth = 2
RFC2833PayloadType = 96

[WEB Params]
LogoWidth = '339'

[SIP Params]

MAXDIGITS = 30
TIMEBETWEENDIGITS = 4
TIMEFORREORDERTONE = 5
REGISTRATIONTIME = 3600
ISPROXYUSED = 1
ISREGISTERNEEDED = 1
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
DEFAULTNUMBER = '1000'
SIPGATEWAYNAME = '10.18.0.204'
USERNAME = 'audiocodes'
CNONCE = '0a123bcf'
PASSWORD = 'blah'
RXDTMFOPTION = 3
ISFAXUSED = 1
GWREGISTRATIONNAME = 'audiocodes'
CODERNAME = g711Ulaw64k,20,0,0,0
CODERNAME = g7231,30,1,$$,0
PREFIX = 1200,10.18.0.204,*,0,255
PREFIX = *,*,*,0,255
PSTNPREFIX = *,1,*,*,0
TRUNKGROUP_1 = 0-0/1-23,1100,0
TRUNKGROUP_1 = 0-0/1-23,1200,0
TRUNKGROUP_1 = 1-23,1300,0
TRUNKGROUP_1 = 1-23,1400,0
PROXYIP = 10.18.0.204
TRUNKGROUPSETTINGS = 1,1
TXDTMFOPTION = 4
```

Web Administration Tool

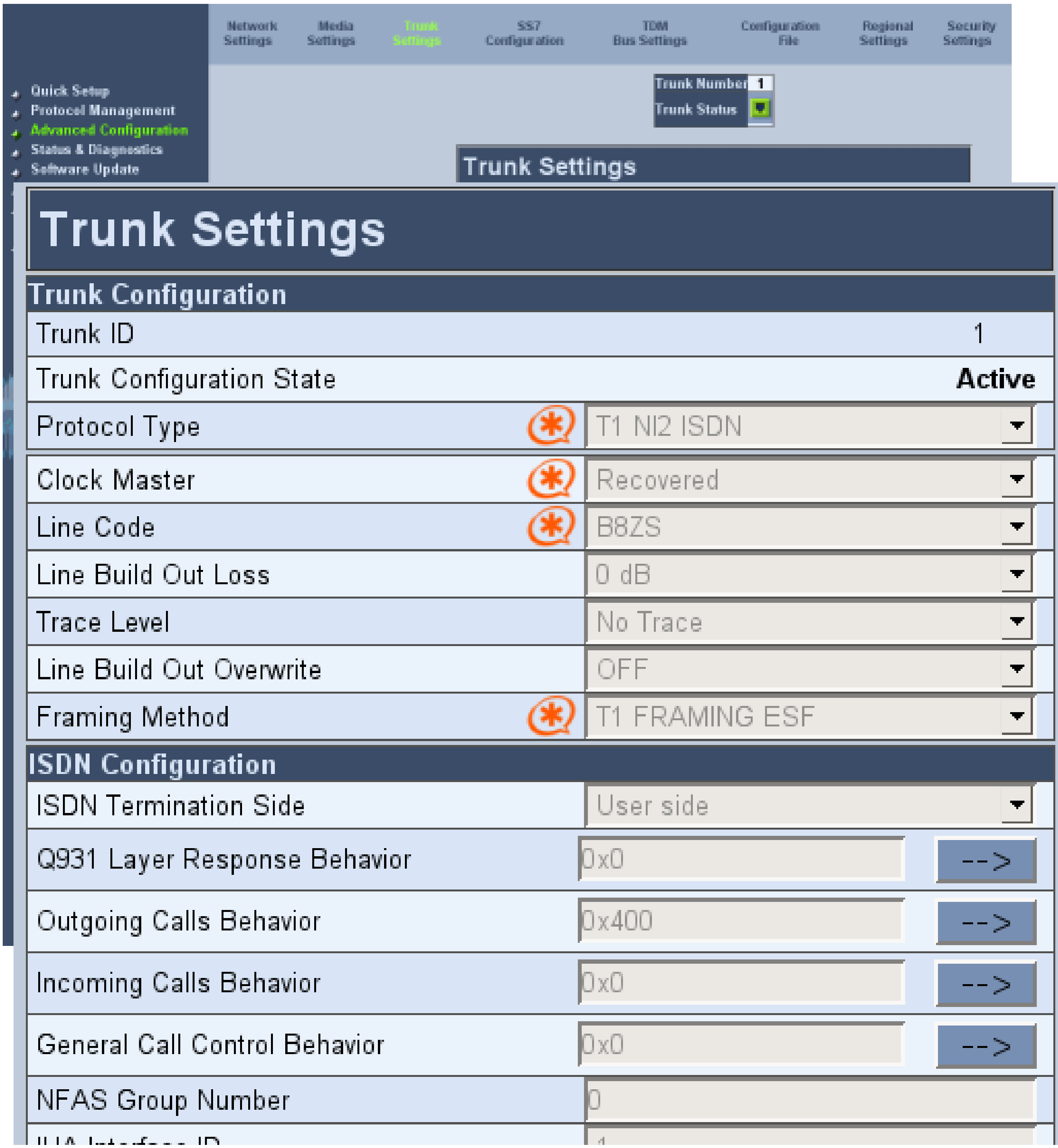
Quick Setup



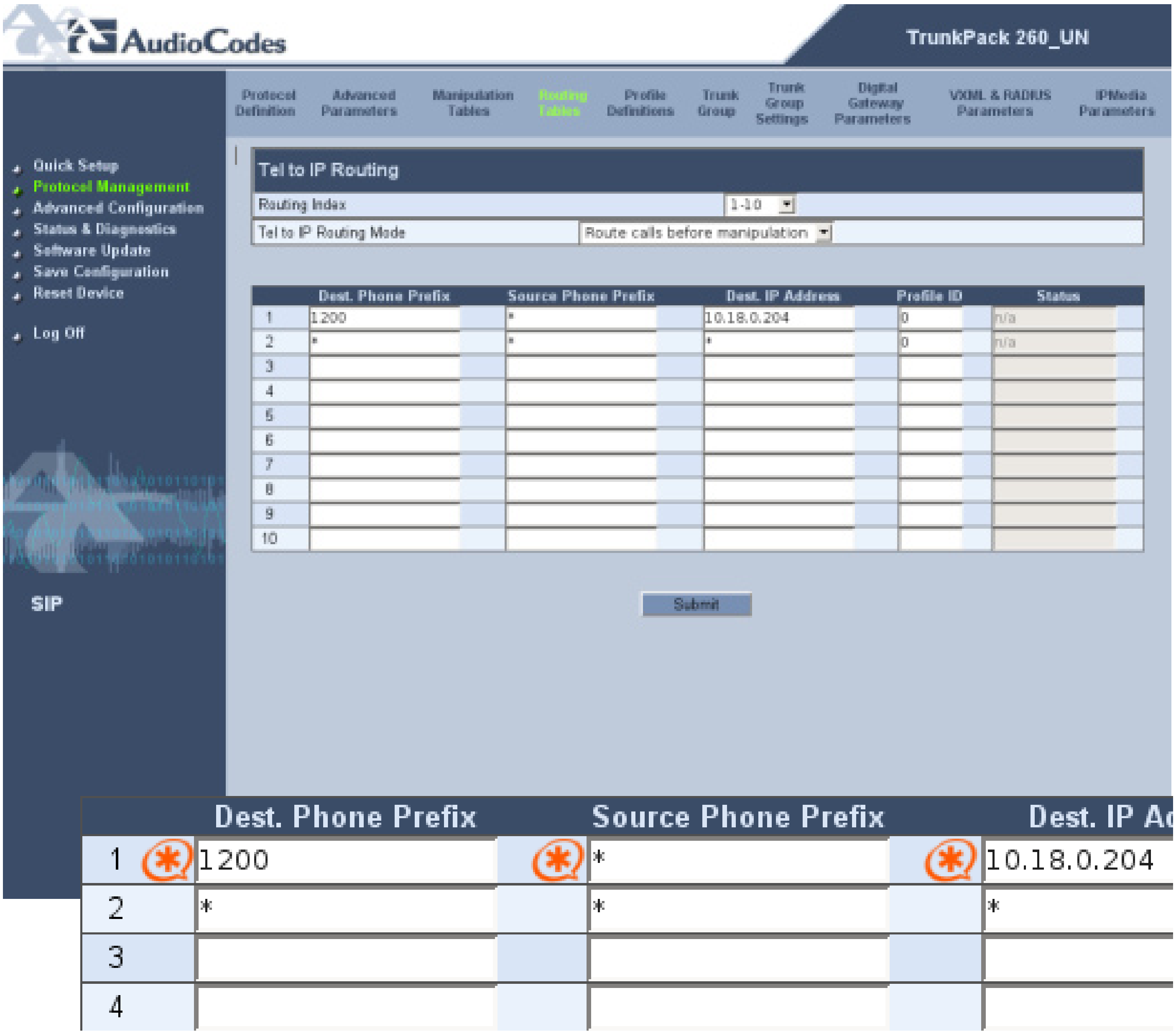
Configuration File

This is where the *.ini configuration files are saved and loaded.

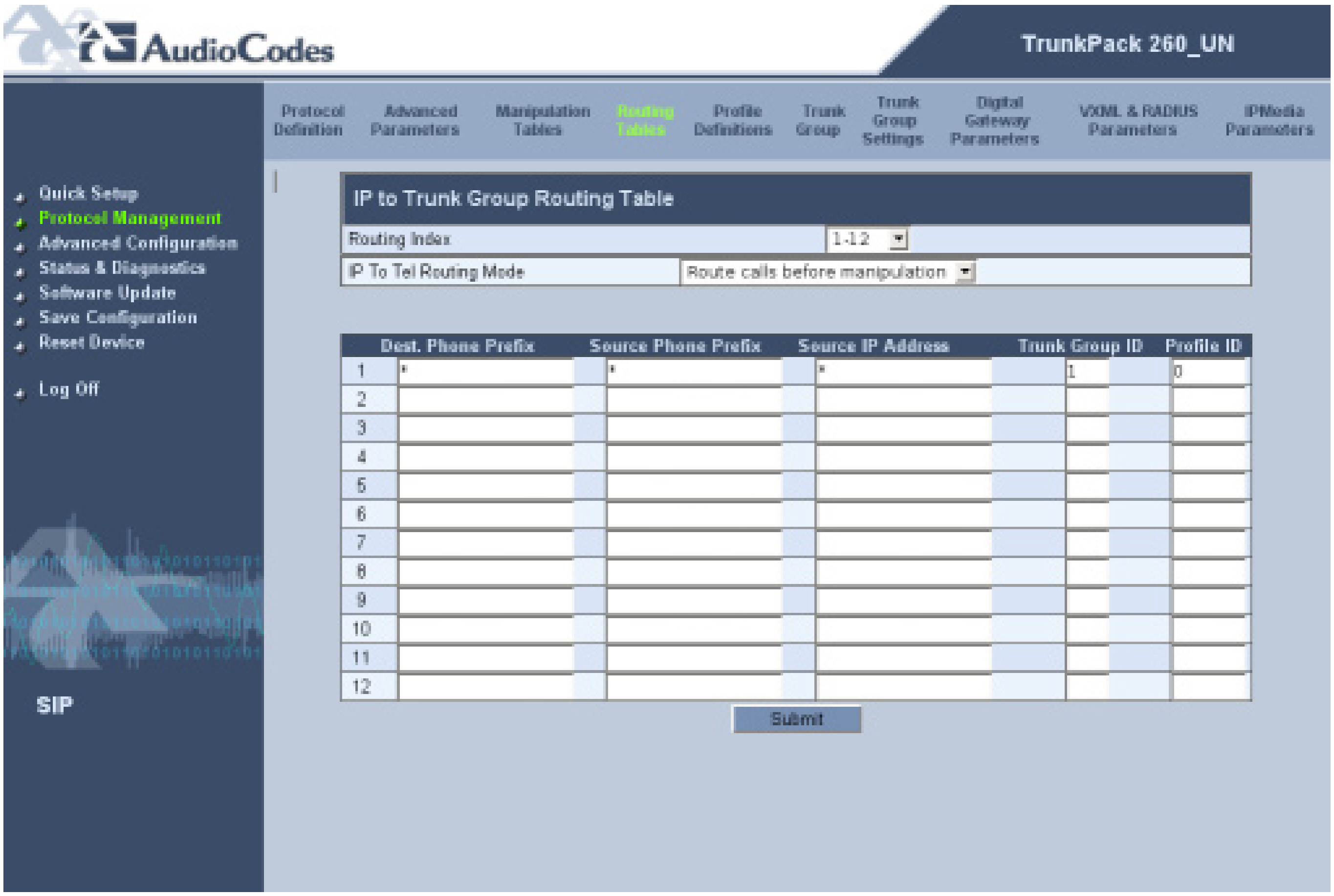
Trunk Settings



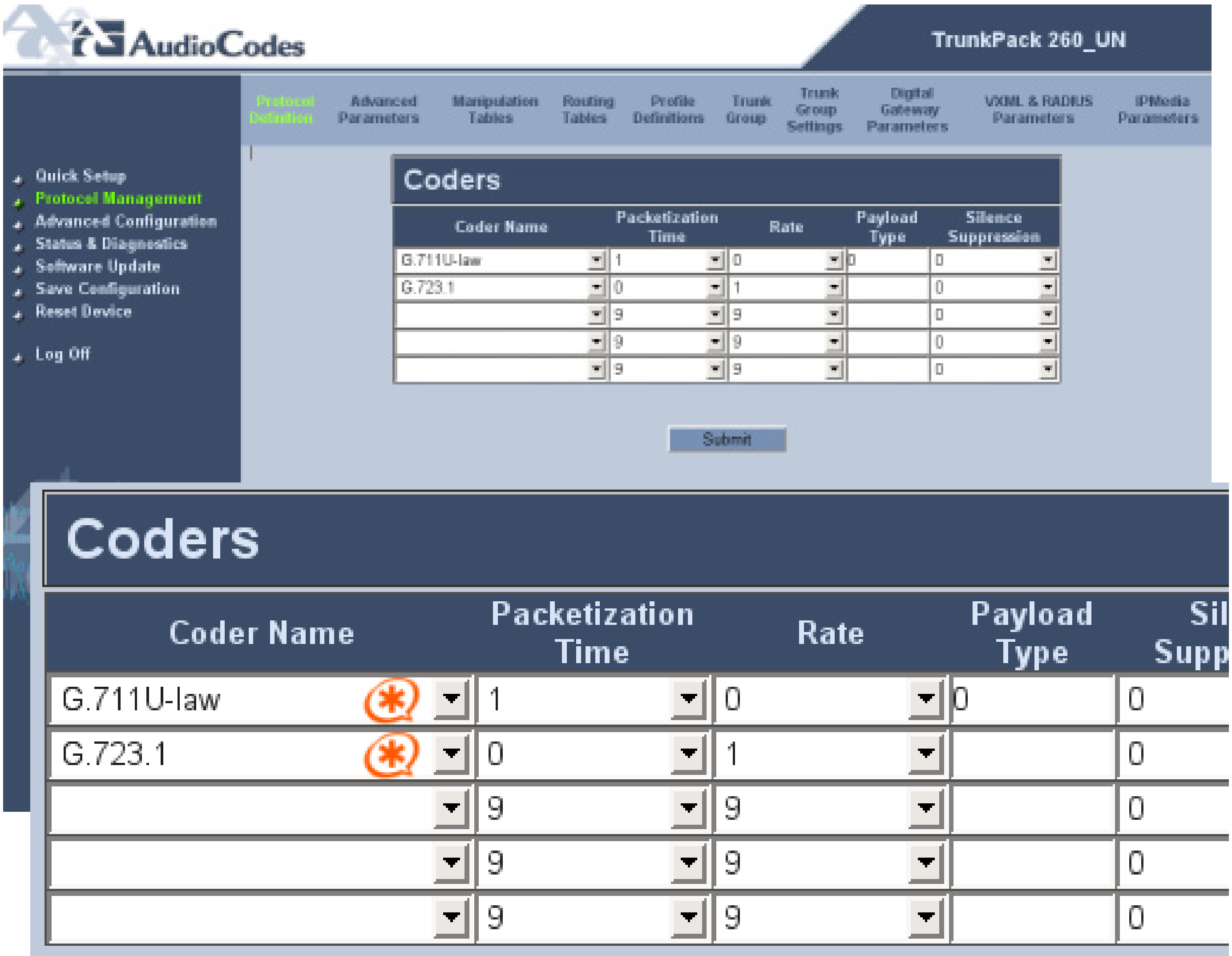
Tel to IP Routing



IP to Trunk Routing



Coders



Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	1	0	0	0
G.723.1	0	1		0
	9	9		0
	9	9		0
	9	9		0

SIP Registration

Description
This test is carried out in order to verify that the Audiocodes gateway can successfully register to the Asterisk PBX.

Procedure
With Asterisk and the TP-260 Gateway configured as outlined before, access the Asterisk CLI, enable sip debugging (via 'sip debug' command) and reboot the TP-260.

Results
As expected, the Gateway is able to register with the Asterisk PBX.

Console Output

```
<-- SIP read from 10.16.3.79:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.16.3.48:5060;branch=z9hG4bK3c2252e6;rport
From: "asterisk" <sip:asterisk@10.16.3.48>;tag=as0445ed1a
To: <sip:audiocodes@10.16.3.79>;tag=1c1918178061
Call-ID: 4d93452d232c765c12cd95f66b5a2023@10.16.3.48
CSeq: 102 OPTIONS
Supported: em,100rel,timer,replaces,path
Allow:REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,S
UBSCRIBE,UPDATE
Content-Type: application/sdp
Content-Length: 176
X-Resources: telchs=1/1;mediachs=0/0

v=0
o=AudiocodesGW 1918183470 1918183342 IN IP4 10.16.3.79
s=Phone-Call
c=IN IP4 10.16.3.79
t=0 0
m=audio 6000 RTP/AVP 2
a=rtpmap:2 g726-32/8000
a=ptime:20
a=sendrecv

--- (11 headers 9 lines)---
Destroying call '4d93452d232c765c12cd95f66b5a2023@10.16.3.48'
deim*CLI>
<-- SIP read from 10.16.3.79:5060:
```

SIP De-Registration

Description
This test is performed in order to verify that after the specified timeout has been reached, the Audiocodes Gateway loses registration from the Asterisk PBX.

Procedure
With the maxexpirey set in sip.conf (ie. ‘maxexpirey=60’), access the Asterisk CLI and wait for the expiration to take place.

Results
As expected, the Audiocodes TP-260 succesfully de-registered from the Asterisk PBX.

Console Output

```
<-- SIP read from 10.16.3.79:5060:
REGISTER sip:10.16.3.48 SIP/2.0
Via: SIP/2.0/UDP 10.16.3.79;branch=z9hG4bKac162228117
Max-Forwards: 70
From: <sip:audiocodes@10.16.3.79>;tag=1c162224230
To: <sip:audiocodes@10.16.3.79>
Call-ID: 185652129211200001156@10.16.3.79
CSeq: 1853 REGISTER
Contact: <sip:audiocodes@10.16.3.79>;expires=60
Supported: em,timer,replaces,path
Allow:REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,S
UBSCRIBE,UPDATE
Authorization:Digestusername=”audiocodes”,realm=”asterisk”,nonce=”284
9aa16”,uri=”sip:10.16.3.48”,algorithm=MD5,response=”a0404b95f21dd152e5b
09451cf678051”
Expires: 60
User-Agent:Audiocodes-Sip-Gateway-TP-260FXS/v.4.80A.014.006
Content-Length: 0


--- (14 headers 0 lines)---
Using latest REGISTER request as basis request
Sending to 10.16.3.79 : 5060 (non-NAT)
Transmitting (no NAT) to 10.16.3.79:5060:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP10.16.3.79;branch=z9hG4bKac162228117;received=10.16.3
.79
```

IP -> PSTN Call Routing

Description
This test is performed in order to verify that the Audiocodes TP-260 correctly routes calls from the SIP trunk to the T1 trunk.

Procedure
Dial IP->PSTN extension (1100) from an analog handset, IP phone, or console, enter a few digits and verify playback.

Results
As expected, Asterisk is able to send a call to the SIP trunk. After being routed by the TP-260 to the T1 trunk, the call is recieved by Asterisk (via TE110p) and the specified audio file is played back.

Console Output

```
herro*CLI> console dial 1100
[Feb 6 01:28:42] WARNING[5401]: chan_oss.c:686 setformat: Unable to re-open DSP device /dev/dsp: No such file or directory
 -- Executing [1100@default:1] Dial("OSS/dsp", "sip/1100@audiocodes") in new stack
    -- Called 1100@audiocodes
    -- Accepting call from '' to '1100' on channel 0/6, span 1
 -- Executing [1100@t1-trunk:1] BackGround("Zap/6-1", "tt-monkeys") in new stack
    -- <Zap/6-1> Playing 'tt-monkeys' (language 'en')
    -- SIP/audiocodes-08b15ed0 answered OSS/dsp
 << Console call has been answered >>
```

IP -> PSTN Call Routing

Description
This test is performed in order to verify that the Audiocodes TP-260 correctly routes calls from the T1 trunk to the SIP trunk.

Procedure
Dial PSTN->IP extension (1200) from an analog handset, IP phone, or console. Enter a few digits and verify playback.

Results
As expected, Asterisk is able to send a call to the SIP trunk. After being routed by the TP-260 to the T1 trunk, the call is recieved by Asterisk (via TE110p) and the specified audio file is played back.

Console Output

```
herro*CLI> console dial 1200
[Feb 6 01:30:28] WARNING[5401]: chan_oss.c:686 setformat: Unable to
re-open DSP device /dev/dsp: No such file or directory
 -- Executing [1200@default:1] Dial("OSS/dsp", "Zap/g1/1200") in new
stack
    -- Requested transfer capability: 0x00 - SPEECH
    -- Called g1/1200
  -- Executing   [1200@sip-trunk:1]      BackGround("SIP/audiocodes-
08b15ed0", "demo-congrats") in new stack
 -- <SIP/audiocodes-08b15ed0> Playing 'demo-congrats' (language
'en')
    -- Zap/1-1 is proceeding passing it to OSS/dsp
[Feb 6 01:30:28] WARNING[5414]: chan_oss.c:984 oss_indicate: Don't
know how to display condition 15 on OSS/dsp
    -- Zap/1-1 answered OSS/dsp
 << Console call has been answered >>
```

Codec Compatibility

Description
This test is carried out in order to verify interoperability between the TP-260 Gateway and the Asterisk PBX using various codecs.

Procedure
Set allow/disallow within sip.conf to allow and restrict the use of codecs with Asterisk. Then specify the codec to be used within the 'Coders' section of the TP-260's web interface.

Results
As expected, the TP-260 is able to utilize all supported codecs (ulaw, alaw, gsm, g.729, g.723.1, g.726).

DTMF Transmission/Reception

Description
This test is performed in order to verify that both Asterisk and the TP-260 handle DTMF correctly.

Procedure
Dial established extensions (1111 and 1212) from an analog handset, IP phone, or console. Enter a few digits and verify NoOp output.

Results
As expected, Asterisk is able to read DTMF sent by the TP-260 gateway, and vice-versa.