

# Audiocodes TP-260

**Form:** Asterisk Interoperability Report

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## **Audiocodes TP-260**

**Asterisk Interoperability Report  
February 2007**

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## Audiocodes TP-260

### Asterisk Interoperability Report

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.

## Table of Contents

Device Summary .....	3
Device Description .....	3
Functionality Testing Overview .....	3
Asterisk Configuration .....	4
Asterisk Configuration continued.....	5
Device Configuration .....	6
Audiocodes BOARD.ini Configuration File.....	6
BOARD.ini (continued) .....	7
Web Administration Tool .....	8
Quick Setup .....	8
Configuration File .....	9
Trunk Settings.....	10
Tel to IP Routing.....	11
IP to Trunk Routing.....	12
Coders .....	13
SIP Registration .....	14
SIP De-Registration .....	15
IP -> PSTN Call Routing .....	16
IP -> PSTN Call Routing .....	17
Codec Compatibility .....	18
DTMF Transmission/Reception .....	18

# Audiocodes TP-260

Form: Asterisk Interoperability Report



## Device Summary

<b>Device Type:</b>	Stand-Alone VoIP Media Gateway
<b>Device Model:</b>	TP-260 SIP
<b>Asterisk Version:</b>	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 successful registration occurs. The extensions.conf file contains the dial plan logic portion of the test setup.

<b>sip.conf</b>
<pre>[audiocodes] context=sip-trunk dtmfmode=rfc2833 host=dynamic qualify=yes nat=never type=peer  [audiocodes] context=sip-trunk dtmfmode=rfc2833 host=dynamic type=user</pre>
<b>extensions.conf</b>
<pre>[default] ;Dial Audiocodes from sip to test IP-&gt;PSTN exten =&gt; 1100,1,Dial(sip/\${EXTEN}@audiocodes) exten =&gt; 1111,1,Dial(sip/\${EXTEN}@audiocodes) ;Dial zap group 1 to test PSTN-&gt;IP exten =&gt; 1200,1,Dial(Zap/g1/\${EXTEN}) exten =&gt; 1212,1,Dial(Zap/g1/\${EXTEN})  [sip-trunk] ;Playback tt-weasels when PSTN-&gt;IP successful exten =&gt; 1200,1,Background(demo-congrats) exten =&gt; 1200,n,Hangup ;Read DTMF Test (PSTN-&gt;IP) exten =&gt; 1212,1,Read(testvar) exten =&gt; 1212,n,NoOp(\${testvar}) exten =&gt; 1212,n,Hangup  [t1-trunk] ;Playback tt-monkey when IP-&gt;PSTN successful exten =&gt; 1100,1,Background(tt-monkeys) exten =&gt; 1100,n,Hangup ;Read DTMF Test (IP-&gt;PSTN) exten =&gt; 1111,1,Read(testvar2) exten =&gt; 1111,n,NoOp(\${testvar2}) exten =&gt; 1111,n,Hangup</pre>

## 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, $
```

# Audiocodes TP-260

Form: Asterisk Interoperability Report



## 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



The screenshot shows the 'Quick Setup' configuration page in the AudioCodes TrunkPack 260\_UN web administration tool. The page is divided into several sections: IP Configuration, SIP Parameters, and Tables. The IP Configuration section includes fields for IP Address (10.18.0.25), NAT IP Address (0.0.0.0), Subnet Mask (255.255.252.0), and Default Gateway IP Address (10.18.0.1). The SIP Parameters section includes Gateway Name (10.18.0.204), Working with Proxy (Yes), Proxy IP Address (10.18.0.204), Proxy Name, and Enable Registration (Enable). The Tables section includes Coders Table, Tel to IP Routing Table, and Trunk Group Table, each with a right-pointing arrow button.

IP Configuration	
IP Address	10.18.0.25
NAT IP Address	0.0.0.0
Subnet Mask	255.255.252.0
Default Gateway IP Address	10.18.0.1

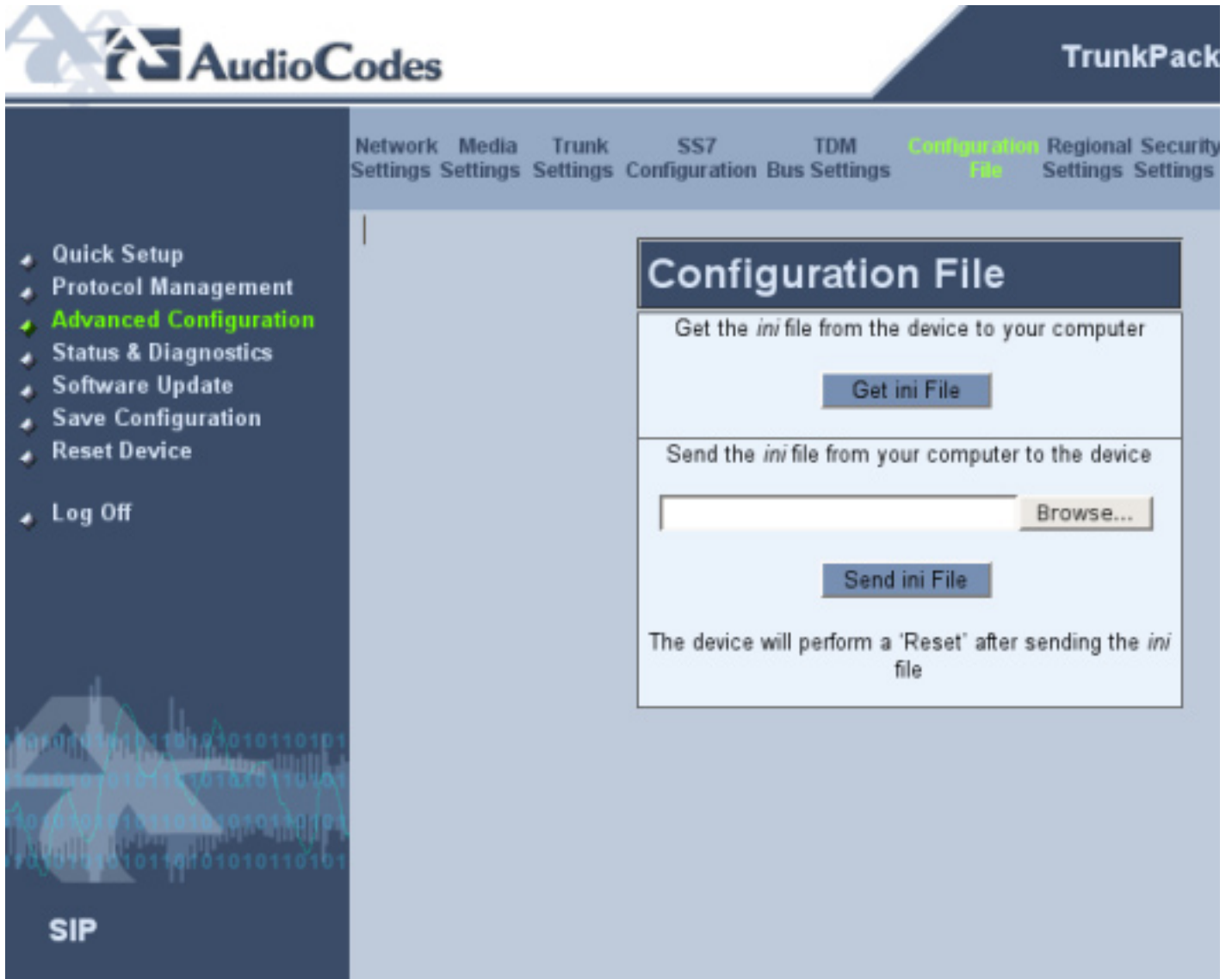
SIP Parameters	
Gateway Name	10.18.0.204
Working with Proxy	Yes
Proxy IP Address	10.18.0.204
Proxy Name	
Enable Registration	Enable

Tables	
Coders Table	-->
Tel to IP Routing Table	-->
Trunk Group Table	-->



## Configuration File

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



The screenshot displays the AudioCodes TrunkPack web interface. The top navigation bar includes 'Network Settings', 'Media Settings', 'Trunk Settings', 'SS7 Configuration', 'TDM Bus Settings', 'Configuration File' (highlighted in green), 'Regional Settings', and 'Security Settings'. A left sidebar menu lists: 'Quick Setup', 'Protocol Management', 'Advanced Configuration' (highlighted in green), 'Status & Diagnostics', 'Software Update', 'Save Configuration', 'Reset Device', and 'Log Off'. The main content area is titled 'Configuration File' and contains two sections: 'Get the ini file from the device to your computer' with a 'Get ini File' button, and 'Send the ini file from your computer to the device' with a text input field, a 'Browse...' button, and a 'Send ini File' button. A note at the bottom states: 'The device will perform a 'Reset' after sending the ini file'. The bottom left corner of the interface features a 'SIP' label and a decorative graphic of binary code.

## Trunk Settings

Network Settings
Media Settings
Trunk Settings
SS7 Configuration
TDM Bus Settings
Configuration File
Regional Settings
Security Settings

- Quick Setup
- Protocol Management
- Advanced Configuration
- Status & Diagnostics
- Software Update

Trunk Number **1**

Trunk Status ▼

Trunk Settings

### Trunk Settings

Trunk Configuration

Trunk ID		1	
Trunk Configuration State		<b>Active</b>	
Protocol Type		T1 NI2 ISDN	▼
Clock Master		Recovered	▼
Line Code		B8ZS	▼
Line Build Out Loss		0 dB	▼
Trace Level		No Trace	▼
Line Build Out Overwrite		OFF	▼
Framing Method		T1 FRAMING ESF	▼




ISDN Configuration

ISDN Termination Side		User side	
Q931 Layer Response Behavior		0x0	-->
Outgoing Calls Behavior		0x400	-->
Incoming Calls Behavior		0x0	-->
General Call Control Behavior		0x0	-->
NFAS Group Number		0	
ISDN Termination ID		1	

## Tel to IP Routing



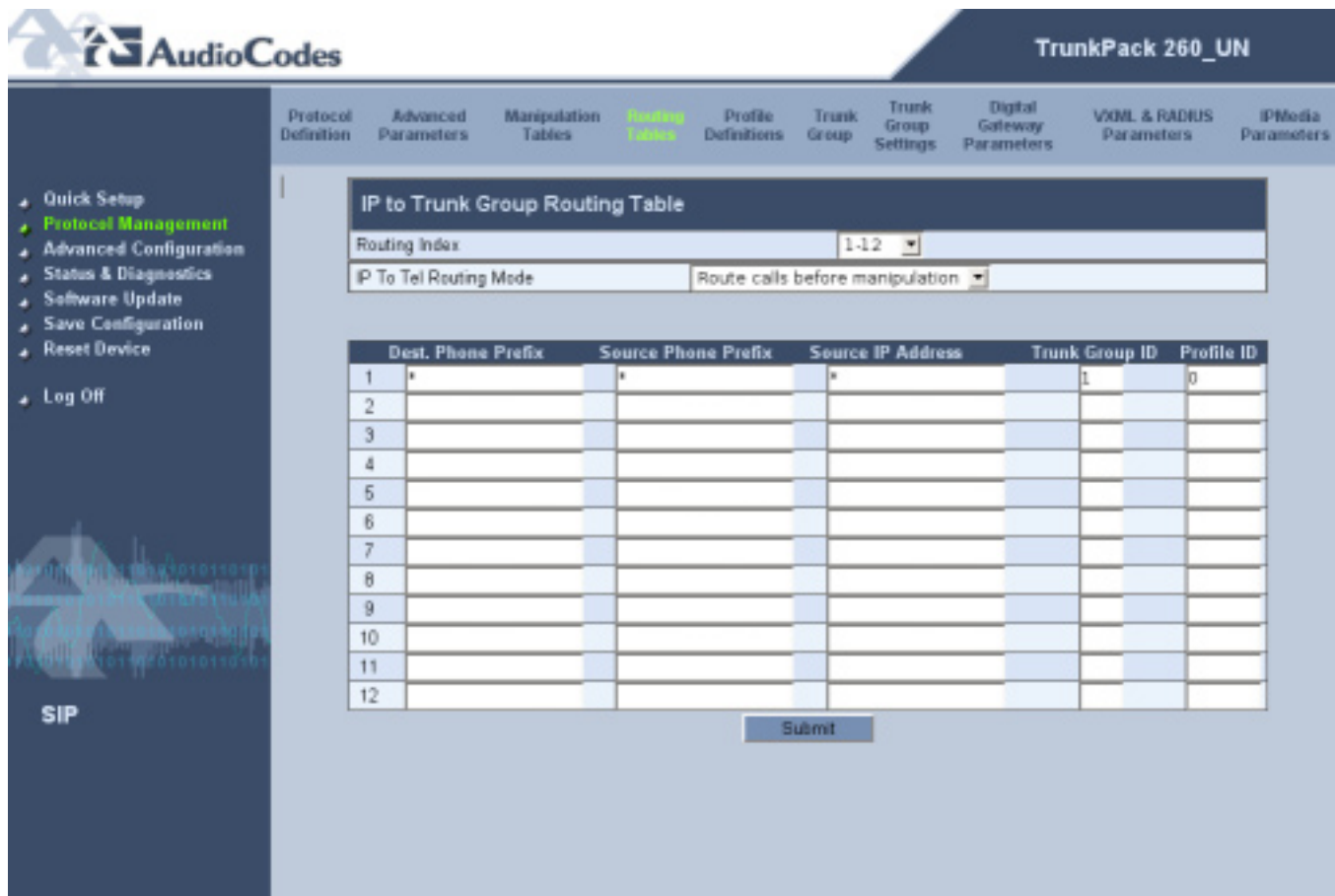
The screenshot shows the configuration page for 'Tel to IP Routing' in the AudioCodes TrunkPack 260\_UN interface. The page includes a navigation menu on the left with options like 'Quick Setup', 'Protocol Management', and 'Log Off'. The main content area has tabs for 'Protocol Definition', 'Advanced Parameters', 'Manipulation Tables', 'Routing Tables', 'Profile Definitions', 'Trunk Group', 'Trunk Group Settings', 'Digital Gateway Parameters', 'VXML & RADIUS Parameters', and 'IPMedia Parameters'. The 'Routing Tables' tab is active, showing a table with columns for 'Dest. Phone Prefix', 'Source Phone Prefix', 'Dest. IP Address', 'Profile ID', and 'Status'. The table contains 10 rows, with the first row populated with '1200', '\*', '10.18.0.204', '0', and 'n/a'. A 'Submit' button is located at the bottom of the table.

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Ad
1	 1200	 *	 10.18.0.204
2	*	*	*
3			
4			

# Audiocodes TP-260

Form: Asterisk Interoperability Report

## IP to Trunk Routing



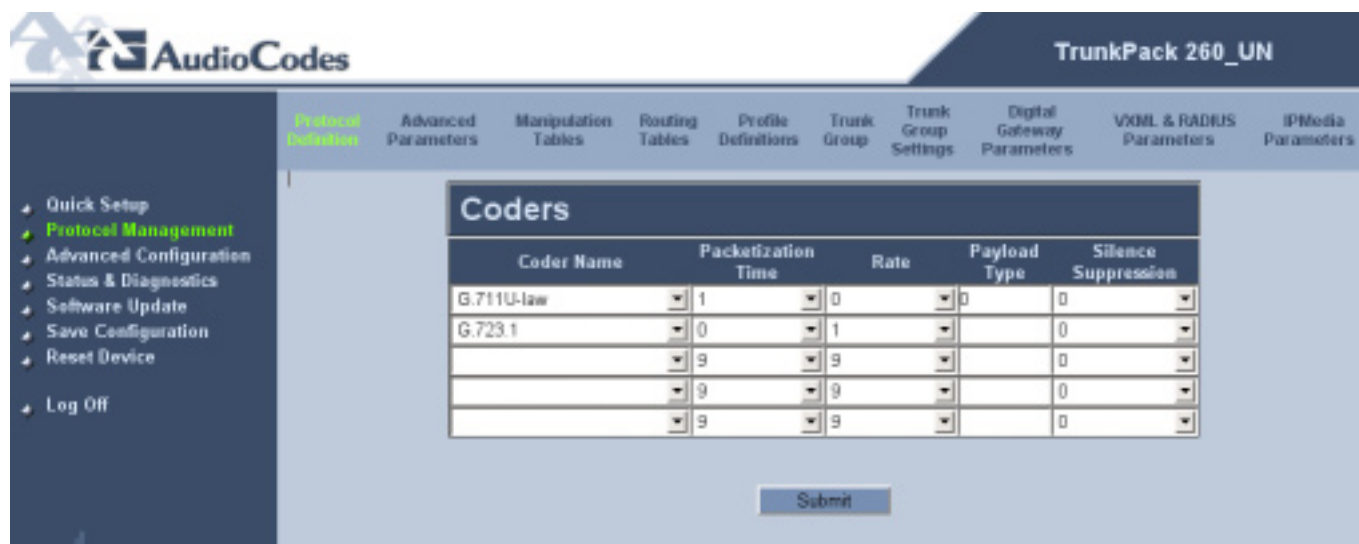
The screenshot shows the AudioCodes TrunkPack 260 configuration interface. The top navigation bar includes tabs for Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables (highlighted), Profile Definitions, Trunk Group, Trunk Group Settings, Digital Gateway Parameters, VOXML & RADIUS Parameters, and IPMedia Parameters. A left sidebar contains a menu with options: Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Save Configuration, Reset Device, and Log Off. The main content area is titled "IP to Trunk Group Routing Table" and features a "Routing Index" dropdown set to "1-12" and an "IP To Tel Routing Mode" dropdown set to "Route calls before manipulation". Below this is a table with 12 rows and 5 columns: Dest. Phone Prefix, Source Phone Prefix, Source IP Address, Trunk Group ID, and Profile ID. The first row contains asterisks in the first three columns, and the Trunk Group ID is 1 and Profile ID is 0. A "Submit" button is located at the bottom of the table.

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	Profile ID
1	*	*	*	1	0
2					
3					
4					
5					
6					
7					
8					
9					
10					
11					
12					

# Audiocodes TP-260

Form: Asterisk Interoperability Report



## Coders



The screenshot shows the Asterisk web interface for the 'TrunkPack 260\_UN' configuration. The 'Coders' section is active, displaying a table with columns for Coder Name, Packetization Time, Rate, Payload Type, and Silence Suppression. The table contains five rows, with the first two rows populated with 'G.711U-law' and 'G.723.1' respectively. A 'Submit' button is located below the table.

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

### Coders

Coder Name	Packetization Time	Rate	Payload Type	Silence Supp
G.711U-law 	1	0	0	0
G.723.1 	0	1	0	0
	9	9	0	0
	9	9	0	0
	9	9	0	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 successfully 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-260 FXS/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 received 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 received 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.