

Dialogic[®] Brooktrout[®] SR140 Fax Software with AudioCodes MediaPack[™] Gateway

Installation and Configuration Integration Note

IMPORTANT NOTE

This document is not to be shared with or disseminated to other third parties, in whole or in part, without prior written permission from Dialogic. To seek such permission, please contact your Dialogic Sales Representative.

September 2009

64-0600-07

www.dialogic.com

Copyright and Legal Notice

Copyright © 2009 Dialogic Corporation. All Rights Reserved. You may not reproduce this document in whole or in part without permission in writing from Dialogic Corporation at the address provided below.

All contents of this document are furnished for informational use only and are subject to change without notice and do not represent a commitment on the part of Dialogic Corporation or its subsidiaries ("Dialogic"). Reasonable effort is made to ensure the accuracy of the information contained in the document. However, Dialogic does not warrant the accuracy of this information and cannot accept responsibility for errors, inaccuracies or omissions that may be contained in this document.

INFORMATION IN THIS DOCUMENT IS PROVIDED IN CONNECTION WITH DIALOGIC[®] PRODUCTS. NO LICENSE, EXPRESS OR IMPLIED, BY ESTOPPEL OR OTHERWISE, TO ANY INTELLECTUAL PROPERTY RIGHTS IS GRANTED BY THIS DOCUMENT. EXCEPT AS PROVIDED IN A SIGNED AGREEMENT BETWEEN YOU AND DIALOGIC, DIALOGIC ASSUMES NO LIABILITY WHATSOEVER, AND DIALOGIC DISCLAIMS ANY EXPRESS OR IMPLIED WARRANTY, RELATING TO SALE AND/OR USE OF DIALOGIC PRODUCTS INCLUDING LIABILITY OR WARRANTIES RELATING TO FITNESS FOR A PARTICULAR PURPOSE, MERCHANTABILITY, OR INFRINGEMENT OF ANY INTELLECTUAL PROPERTY RIGHT OF A THIRD PARTY.

Dialogic products are not intended for use in medical, life saving, life sustaining, critical control or safety systems, or in nuclear facility applications.

Due to differing national regulations and approval requirements, certain Dialogic products may be suitable for use only in specific countries, and thus may not function properly in other countries. You are responsible for ensuring that your use of such products occurs only in the countries where such use is suitable. For information on specific products, contact Dialogic Corporation at the address indicated below or on the web at www.dialogic.com.

It is possible that the use or implementation of any one of the concepts, applications, or ideas described in this document, in marketing collateral produced by or on web pages maintained by Dialogic may infringe one or more patents or other intellectual property rights owned by third parties. Dialogic does not provide any intellectual property licenses with the sale of Dialogic products other than a license to use such product in accordance with intellectual property owned or validly licensed by Dialogic and no such licenses are provided except pursuant to a signed agreement with Dialogic. More detailed information about such intellectual property is available from Dialogic's legal department at 9800 Cavendish Blvd., 5th Floor, Montreal, Quebec, Canada H4M 2V9. Dialogic encourages all users of its products to procure all necessary intellectual property infringement and disclaims any responsibility related thereto. These intellectual property licenses may differ from country to country and it is the responsibility of those who develop the concepts or applications to be aware of and comply with different national license requirements.

Any use case(s) shown and/or described herein represent one or more examples of the various ways, scenarios or environments in which Dialogic® products can be used. Such use case(s) are non-limiting and do not represent recommendations of Dialogic as to whether or how to use Dialogic products.

Dialogic, Dialogic Pro, Brooktrout, Diva, Cantata, SnowShore, Eicon, Eicon Networks, NMS Communications, NMS (stylized), Eiconcard, SIPcontrol, Diva ISDN, TruFax, Exnet, EXS, SwitchKit, N20, Making Innovation Thrive, Connecting to Growth, Video is the New Voice, Fusion, Vision, PacketMedia, NaturalAccess, NaturalCallControl, NaturalConference, NaturalFax and Shiva, among others as well as related logos, are either registered trademarks or trademarks of Dialogic Corporation or its subsidiaries. Dialogic's trademarks may be used publicly only with permission from Dialogic. Such permission may only be granted by Dialogic's legal department at 9800 Cavendish Blvd., 5th Floor, Montreal, Quebec, Canada H4M 2V9. Any authorized use of Dialogic's trademarks will be subject to full respect of the trademark guidelines published by Dialogic from time to time and any use of Dialogic's trademarks requires proper acknowledgement.

The names of actual companies and products mentioned herein are the trademarks of their respective owners.

This document discusses one or more open source products, systems and/or releases. Dialogic is not responsible for your decision to use open source in connection with Dialogic products (including without limitation those referred to herein), nor is Dialogic responsible for any present or future effects such usage might have, including without limitation effects on your products, your business, or your intellectual property rights.

Any use case(s) shown and/or described herein represent one or more examples of the various ways, scenarios or environments in which Dialogic products can be used. Such use case(s) are non-limiting and do not represent recommendations of Dialogic as to whether or how to use Dialogic products.

1. Scope

This document is intended as a general guide for configuring a basic installation of the *AudioCodes MediaPack*TM *Gateway* for use with Dialogic[®] Brooktrout[®] SR140 Fax over IP (FoIP) software platform. The interoperability includes *SIP* call control and T.38/T.30 media.

This document is not intended to be comprehensive, and thus should not and does not replace the manufacturer's detailed configuration documentation. Users of this document should already have a general knowledge of how to install and configure the *AudioCodes MediaPack*TM *Gateway.*

The sample configuration shown and/or referred in the subsequent sections was used for lab validation testing by Dialogic. Therefore, it is quite possible that the sample configuration will not match an exact configuration or versions that would be present in a deployed environment. However, the sample configuration does provide a possible starting point to work with the equipment vendor for configuring your device. Please consult the appropriate manufacturer's documentation for details on setting up your specific end user configuration.

2. Configuration Details

The following systems were used for the sample configuration described in the document.

2.1 Gateway

Vendor	AudioCodes
Model	MediaPack [™] MP-114
Software Version	Firmware 5.60A.025.005
PSTN Device	Dialogic® Brooktrout® TR1034 Analog Fax Board
Protocol to PSTN Device	Analog
IP Device	Dialogic® Brooktrout® SR140 Fax Software
Additional Notes	All Defaults were used except SIP transport was changed to UDP

For ease of reference, the Dialogic[®] Brooktrout[®] SR140 Fax Software and Dialogic[®] Brooktrout[®] TR1034 Fax Boards will sometimes be denoted herein, respectively, as SR140 and TR1034, and the *AudioCodes MediaPack[™] MP-114* will be denoted herein as MediaPack[™] or *MP-114*, or some other form thereof. Also, all mentions of SDK herein refer to the Dialogic[®] Brooktrout[®] SDK.

2.2 Dialogic[®] Brooktrout[®] SR140 Fax Software

Vendor	Dialogic
Model	Dialogic® Brooktrout® SR140 Fax Software
Software Version	SDK 6.1.0
Protocol to Gateway or Call Manager	SIP
callctrl.cfg file	All defaults

2.3 Dialogic[®] Brooktrout[®] TR1034 Fax Board

Vendor	Dialogic
PSTN Device	Dialogic® Brooktrout® TR1034 Fax Board
Software Version	SDK 6.1.0
Protocol to PSTN Device	Analog
callctrl.cfg file	All defaults

2.4 Network System Configuration

The diagram below details the sample configuration used in connection with this document.



Diagram Notes:

- SR140 Fax Server = Fax Server including Dialogic[®] Brooktrout[®] SR140 Fax Software and 3rd party fax application
- TR1034 Fax Server = Fax Server including Dialogic[®] Brooktrout[®] TR1034 Fax Board and 3rd party fax application

3. Prerequisites

No special requirements to note.

4. Summary of Limitations

TCP cannot be used as a SIP Transport Type because the SR140 software does not support SIP over TCP. This parameter should be set to UDP on the MP-114 SIP General Parameters screen.

5. Dialogic[®] Brooktrout[®] SR140 Fax Software Setup Notes

For the sample test configuration, the SR140 was configured using the default values, consult the Dialogic[®] Brooktrout[®] Fax Products Installation and Configuration Guide for details.

The Installation and Configuration Guide for SDK 6.1.x is available from the site below:

http://www.dialogic.com/manuals/brooktrout/default.htm

6. Dialogic[®] Brooktrout[®] TR1034 Fax PSTN Setup Notes

For the sample test configuration, the TR1034 was configured using the default values, consult the Dialogic[®] Brooktrout[®] Fax Products Installation and Configuration Guide for details.

7. AudioCodes MediaPack[™] Gateway Setup Notes

7.1 AudioCodes MP-114 FXS Configuration

For the sample test configuration, the following instructions were used to configure the AudioCodes MediaPack MP-114 FXS for T.38 faxing. For more information on the parameters described below, refer to the AudioCodes User's Guide.

1. Access board's web interface (<u>http://xx.xx.xx.xx</u>)

2. Default password and username: Admin



3. Configure "Endpoint Phone Number" page

- a. This is the defined number used to identify the port
 - i. Example FXS configuration:
 - 1. 'Channel' # 1
 - a. has a 'Phone Number' of 1001
 - b. is associated with 'Hunt Group ID' # 1
 - i. This ID is used for IP->Tel Routing
 - c. uses the 'Profile ID' of '0'
 - i. ID # '0' is the default setting allowing the port to use the global settings defined on the gateway

Search		Channel(s)	Phone Number	Hunt Group ID	Profile ID
Basic OFull	1	1	1001	1	0
Network Settings	2				
Protocol Configuration	3				
Protocol Definition	4				
Manipulation Tables Bouting Tables	5				
Profile Definitions	6				
Endpoint Settings	7				
EndPoint Number	8				

- 4. Configure "Hunt Group Settings" page
 - a. Use 'By Dest Phone Number' to have the gateway route the call to the specific port associated with the phone number configured in step 3
 - i. Example FXS configuration :
 - 1. Configure 'Hunt Group ID' # 1 (for the endpoint 1 in step 3)
 - a. Select the 'Channel Select Mode' for IP->Tel Routing
 - b. Select the 'Registration Mode' (configure only if registration is required)

enarios Search				Advanced Paramet
Basic O Full	Routi	ng Index	1-12 💌	
Network Settings				
Protocol Configuration		Hunt Group ID	Channel Select Mode	Registration Mode
Protocol Definition	1	1	By Dest Phone Number	
Manipulation Tables	2		×	~
* Profile Definitions	3		×	×
Endpoint Settings	4			
Bendpoint Number	5			
Hunt Group Settings	6			
Advanced Applications	7			
	8			
	9			
	10			
	11			
	12			

- 5. Configure "IP to Trunk Group Routing" page
 - a. This table will route inbound traffic from the IP side to the appropriate 'Hunt Group ID'
 - i. Example FXS configuration:
 - 1. This rule will route ANY incoming traffic to 'Hunt Group ID' # 1 (for the endpoint 1 in step 3)

enarios Search									Adv	anced Parameter
			-							
Basic OFull			Routi	ng Index	1-12 💌		-			
Network Settings			IP To	Tel Routing Mode	Route calls before manipula	ation 🔛	1			
Media Settings		Dest. Phone Pre	efix	Source Phone Prefix	Source IP Address		->	Hunt	Group ID	IP Profile ID
Protocol Definition	1	*		*	*			1		0
Manipulation Tables	2									
Routing Tables	3								_	
IP to Truck Group Routing	4					_			_	
Profile Definitions	5					-			_	
Endpoint Settings	6					-		-	_	
Hunt/IP Group	7					-			_	
Advanced Applications	8					-		-	_	-
	9					_			_	
	10					_	-	_		
	10					_				
	11									

- 6. Configure "Tel to IP Routing" page
 - a. This table is used to route traffic from the FXS ports to a specific IP address
 - i. Example FXS configuration:
 - 1. Index # 1 will route the call to IP address '172.22.2.30' if the FXS user dials '2001'
 - 2. Index # 2 will route the call to IP address '172.22.1.76' if the FXS user dials '7049'

Configuration Management Status & Diagnostics	Tel to IP Routing						Ådua	pood Doromotor List
Scenarios Search		-				-		iscur uruneter Era
Basic OFull	9	Routing I	ndex		1-10	~		
Metwork Settings		Tel To IP	Routin	ig Mode	Rout	e calls bef	ore manipulation 🔛	
Media Settings								
Protocol Configuration E Protocol Definition	Dest. Phone F	Prefix Source Phone Prefi	× ->	Dest. IP Address	Transport Type	Dest. IPGroup	IP Profile ID	Stati
Manipulation Tables			_			ID		
Routing Tables	1 2001	*		172.22.2.30	Not Configured	×	0	n/a
Tel to IP Routing	2 7049	*		172.22.1.76	Not Configured 💌	~	0	n/a
Profile Definitions	3				Not Configured 💌			
Endpoint Settings	4				Not Configured 🔽	~		
* Endpoint Number	5				Not Configured 💌	~		
Advanced Applications	6				Not Configured 💌	~		
	7				Not Configured 🔽			
	8				Not Configured 💌			
	9				Not Configured 💌			
	10				Not Configured	~		

7. Configure "Coders" page

- i. Select the coders that are supported by the IP side
- ii. Do not select 'T.38' as a coder

	Coder Nar	ne	Packetiza	tion Time	Ra	ate	Payload Type	Silence Supp	ression
Basic OFull	G.711U-law	~	20	 Image: A start of the start of	64		0	Disabled	~
Network Settings	G.711A-law	~	20	~	64	~	8	Disabled	~
Protocol Configuration	G.729	~	20	~	8	~	18	Disabled	~
E Protocol Definition		~				~			
SIP General Parameters		~				V			~
Coders DTMR & Dialing Coders									

- 8. Configure "SIP General Parameters" page
 - a. To activate the gateway to use T.38 for fax calls, configure 'Fax Signaling Method' to 'T.38 Relay'

cenarios Search				Advanced Parame
	▼ SIP General			
Basic O Full	PRACK Mode	Supported	~	
Network Settings	Channel Select Mode	By Dest Phone Number	~	
Media Settings Protocol Configuration Protocol Definition SIP General Parameters	Enable Early Media	Disable	~	
	Session-Expires Time	0		
	Minimum Session-Expires	90		
SIP General Parameters	Session Expires Method	Re-INVITE	~	
Proxy & Registration	Asserted Identity Mode	Disabled	~	
Coders	Fax Signaling Method	T.38 Relay	~	
CTMF & Dialing Anipulation Tables Comparison Compa	SIP Transport Type	SIP Transport Type UDP		
	SIP UDP Local Port	5060		
	SIP TCP Local Port	5060		
	SIP TLS Local Port	5061		
	Enable SIPS	Disable	~	
	Enable TCP Connection Reuse	Enable	~	
	SIP Destination Port	5060		
	Enable Remote Party ID	Disable	~	
	Enable History-Info Header	Disable	~	
	Play Ringback Tone to IP	Don't Play	~	
	Play Ringback Tone to Tel	Play According to Early Media	~	
	3xx Behavior	Forward	~	
	Enable Reason Header	Enable	~	

After configuring the AudioCodes MP-114 FXS port for T.38 faxing, the INI file should be similar to the following exported INI file.

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

;Board: MP-114 FXS_FXO ;Serial Number: 1239153 ;Slot Number: 1 ;Software Version: 5.60A.025.005 ;DSP Software Version: 204IM => 560.12 ;Board IP Address: 10.128.28.20 ;Board Subnet Mask: 255.255.252.0 ;Board Default Gateway: 10.128.28.1 ;Ram size: 32M Flash size: 8M ;Num of DSP Cores: 1 Num DSP Channels: 4 ;Profile: NONE

[SYSTEM Params]

SyslogServerIP = 10.15.6.100 VXMLFIleName = "

[BSP Params]

[Analog Params]

FXSLoopCharacteristicsFilename = 'MP11x-02-1-FXS_16KHZ.dat'

[ControlProtocols Params]

[MGCP Params]

[MEGACO Params]

EP_Num_0 = 0 EP_Num_1 = 1 EP_Num_2 = 0 EP_Num_3 = 0 EP_Num_4 = 0

[Voice Engine Params]

FarEndDisconnectSilencePeriod = 60 CallProgressTonesFilename = 'usa_tones_12.dat' ECNLPMode = 1 BrokenConnectionEventTimeout = 3 CNGDetectorMode = 2 RFC2833PayloadType = 101 DTMFDetectorSensitivity = 1

[WEB Params]

[SIP Params]

MAXDIGITS = 4 ISTWOSTAGEDIAL = 0

```
ENABLECURRENTDISCONNECT = 1
ENABLEREVERSALPOLARITY = 1
CDRREPORTLEVEL = 1
GWDEBUGLEVEL = 5
ENABLEVOICEDETECTION = 1
DISCONNECTONBROKENCONNECTION = 0
MWIANALOGLAMP = 1
ENABLEMWI = 1
ISFAXUSED = 1
SUBSCRIPTIONMODE = 1
3XXBEHAVIOR = 1
[IPsec Params]
[SNMP Params]
 *** TABLE DspTemplates ***
This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
 *** TABLE PREFIX ***
[PREFIX]
FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileId,
PREFIX_MeteringCode, PREFIX_DestPort, PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix, PREFIX_DestIPGroupID,
PREFIX_SrcHostPrefix, PREFIX_TransportType, PREFIX_SrcTrunkGroupID;
PREFIX 0 = *, 10.128.28.200, *, 0, 255, 0, -1, , -1, , -1;
PREFIX 1 = 2222, 10.128.16.136, *, 0, 255, 0, -1, , -1, , -1, -1;
PREFIX 2 = 1000, 10.128.16.146, *, 0, 255, 0, -1, , -1, , -1, -1;
[\PREFIX]
 *** TABLE CoderName ***
[CoderName]
FORMAT CoderName_Index = CoderName_Type, CoderName_PacketInterval, CoderName_rate, CoderName_PayloadType,
CoderName_Sce;
CoderName 0 = g711Ulaw64k, 20, 0, 255, 0;
CoderName 1 = g711Alaw64k, 20, 0, 255, 0;
[\CoderName]
 *** TABLE TrunkGroup ***
[TrunkGroup]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, 2001, 0, 255, 255;
TrunkGroup 1 = 0, 255, 2, 2, 2002, 0, 255, 255;
TrunkGroup 2 = 2, 255, 3, 4, 2003, 0, 255, 255;
[\TrunkGroup]
 *** TABLE PstnPrefix ***
```

```
;
```

[PstnPrefix] FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId, PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix; PstnPrefix 0 = 2001, 1, *, *, 0, -1, , ; PstnPrefix 1 = 2002, 1, *, *, 0, -1, , ; PstnPrefix 2 = *, 1, *, *, 0, -1, , ; [\PstnPrefix] *** TABLE TxDtmfOption *** [TxDtmfOption] FORMAT TxDtmfOption_Index = TxDtmfOption_Type; TxDtmfOption 0 = 4;[\TxDtmfOption] *** TABLE TrunkGroupSettings *** [TrunkGroupSettings] FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId, TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode, TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser, TrunkGroupSettings_ServingIPGroup; TrunkGroupSettings 0 = 1, 0, 255, , , -1; TrunkGroupSettings 1 = 2, 2, 255, , , -1; [\TrunkGroupSettings] *** TABLE TargetOfChannel *** [TargetOfChannel] FORMAT TargetOfChannel_Index = TargetOfChannel_Destination, TargetOfChannel_Type; TargetOfChannel 2 = 1111, 1; TargetOfChannel 3 = 1111, 1; [\TargetOfChannel] *** TABLE ProxySet *** [ProxySet] FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap; ProxySet 0 = 0, 60, 0, 0;[\ProxySet]

7.2 AudioCodes MP-114 FXO Configuration

For the sample test configuration, the following instructions were used to configure the AudioCodes MP-114 FXO for T.38 faxing. For more information on the parameters described below, refer to the AudioCodes User's Guide.

1. Access board's web interface (http://xx.xx.xx.xx)

🗇 • 🔖 • 🥑 🛞 🏠 🕒 http://172.20.0.226/	Eile	Edit	⊻iew	History	Bookmarks	Tools	Help
	4	- 📫	- (ଟ 😣		http://1	72.20.0.226/

2. Default password and username: Admin

Authei	ntication Required 🛛 👔
?	Enter username and password for "Realm1" at http://172.20.0.226 User Name:
	Admin
	Password:

	Ise Password Mapager to remember this password.
	OK Cancel

- 3. Configure "Endpoint Phone Number" page
 - a. This can be any number (example the number of the POTS line)
 - i. Example FXO configuration:

C.

- 1. 'Channel' #s 1 to 4 will
 - a. have use a 'Phone Number' range of 1001 to 1004
 - b. be associated with 'Hunt Group ID' # 1
 - i. This ID is used for IP->Tel Routing
 - use the 'Profile ID' of '0'
 - i. ID # '0' is the default setting allowing the port to use the global settings defined on the gateway

enarios Search					
	1	Channel(s)	Phone Number	Hunt Group ID	Profile ID
lasic 🔾 Full	1	1-4	1001	1	0
Network Settings	2				
Protocol Configuration	3				
Protocol Definition	4				
Manipulation Tables	5				
Profile Definitions	6				
Endpoint Settings	7				
Endpoint Number	8				

4. Configure "Hunt Group Settings" page

- a. It is recommend to use 'Cyclic Ascending' or 'Cyclic Descending' to have the gateway route the call to the first available POTS line within the group configured in step 3
 - i. Example FXO configuration:
 - 1. Configure 'Hunt Group ID' # 1 (for the endpoints 1-4 in step 3)
 - a. Select the 'Channel Select Mode' for IP->Tel Routing
 - b. Select the 'Registration Mode' (configure only if registration is required)

Configuration Management Status	Hunt Group Set	tings		
Scenarios Search				Advanced Parameter List
Basic OFull	- Rout	ing Index	1.12	
Network Settings Media Settings		•		V
Protocol Configuration		Hunt Group ID	Channel Select Mode	Registration Mode
Protocol Definition	1	1	Cyclic Ascending	
Manipulation Tables	2			
Profile Definitions	3			
#@Endpoint Settings	4		×	×
Barty Group	5			
Hunt Group Settings	6			
Advanced Applications	7		×	
	8		×	
	9			
	10			
	11			
	12			

- 5. Configure "IP to Trunk Group Routing" page
 - a. This table will route inbound traffic from the IP side to the appropriate 'Hunt Group ID'
 i. Example FXO configuration:
 - This rule will route ANY incoming traffic to 'Hunt Group ID' # 1 (for the endpoints 1-4 in step 3)

Scenarios Search							Adv	anced Parameter Li
		•						
Basic OFull			Routing Index		1-12 💌			
* Network Settings			P To Tel Routing M	ode	Route calls before manipulation			
Media Settings		Dest. Phone Prefix	Sourc	e Phone Prefix	Source IP Address	->	Hunt Group ID	IP Profile ID
Protocol Definition	1	*			*		1	0
Manipulation Tables	2							
Tel to IR Pouting	3					1		
IP to Trunk Group Routing	4							
Profile Definitions	5							
Endpoint Settings Endpoint Number	6		1					
Hunt/IP Group	7							
Advanced Applications	8							
	9							
	10							
	11							
	10							

6. Configure "Tel to IP Routing" page

- a. This table is used to route traffic from the FXO ports to a specific IP address
 - i. Example FXO configuration:
 - 1. Index # 1 will route the call to IP address '172.22.2.30' if the FXO user dials '2001'
 - 2. Index # 2 will route the call to IP address '172.22.1.76' if the FXO user dials '7049'

Scenarios Search		Advar					ced Parameter List 🗨		
Basic O Full	5	▼ Routing Ind Tel To IP Ro	ex outin	a Mode	1-1 Ro	0 💌 .te calls b	efore r	nanipulation	
Media Settings Protocol Configuration Protocol Definition	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Transport Type	Dest IPGro	up	IP Profile ID	Stat
Manipulation Tables	1 2001	*		172.22.2.30	Not Configured		0		n/a
Tel to IP Routing	2 7049	*	1	172.22.1.76	Not Configured		0		n/a
IP to Trunk Group Routing Profile Definitions	3		1		Not Configured				
Endpoint Settings	4		1		Not Configured				
* Endpoint Number	5		1		Not Configured				
Advanced Applications	6		1		Not Configured				
	7		1		Not Configured				
	8				Not Configured	-			
	9		1		Not Configured				
	10		1		Not Configured		i l		

7. Configure "Coders" page

- i. Select the coders that are supported by the IP side
- ii. Do not select 'T.38' as a coder

	Coder Na	me	Packetiza	tion Time	Rat	te	Payload Type	Silence Supp	ression
asic OFull	G.711U-law	~	20	~	64	~	0	Disabled	~
Network Settings	G.711A-law	~	20	~	64			Disabled	~
Protocol Configuration	G.729	~	20	~	8	~	18	Disabled	~
Protocol Definition		~		V					~
SIP General Parameters		~		×		~			~
Coders DTMF & Dialing Manipulation Tables Routing Tables Profile Definitions									

8. Configure "SIP General Parameters" page

a. To activate the gateway to use T.38 for fax calls, configure 'Fax Signaling Method' to 'T.38 Relay'

rios Search				Advanced Paramet
	▼ SIP General			
sic OFull 🕜	PRACK Mode	Supported	~	
letwork Settings	Channel Select Mode	By Dest Phone Number	~	
ledia Settings	Enable Early Media	Disable	~	
rotocol Configuration	Session-Expires Time	0		
Protocol Definition	Minimum Session-Expires	90		
SIP General Parameters	Session Expires Method	Re-INVITE	~	
Proxy & Registration	Asserted Identity Mode	Disabled	~	
Coders	Fax Signaling Method	T.38 Relay	~	2
DTMF & Dialing	SIP Transport Type	UDP	~	
Manipulation Tables	SIP UDP Local Port	5060		
Routing Tables	SIP TCP Local Port	5060		
Profile Definitions	SIP TLS Local Port	5061		
Endpoint Settings	Enable SIPS	Disable	~	
Endpoint Number	Enable TCP Connection Reuse	Enable	~	
Hunt/IP Group	SIP Destination Port	5060		
avanced Applications	Enable Remote Party ID	Disable	~	
	Enable History-Info Header	Disable	~	
	Play Ringback Tone to IP	Don't Play	~	
	Play Ringback Tone to Tel	Play According to Early Media	~	
	3xx Behavior	Forward	~	
	Enable Reason Header	Enable	V	

9. Configure "FXO Settings" page

- a. Depending on the desired method, the FXO ports can be configured for either 'Two Stage' (default) or 'One Stage' dialing
 - i. If two-stage dialing is enabled, the device seizes one of the PSTN/PBX lines without performing any dialing, connects the remote IP user to the PSTN/PBX, and all further signaling (dialing and Call Progress Tones) is performed directly with the PBX without the device's intervention.
 - ii. If one-stage dialing is enabled, the device seizes one of the available lines, and dials the destination phone number received in the INVITE message. Use the parameter IsWaitForDialTone to specify whether the dialing must start after detection of the dial tone or immediately after seizing the line.
 - 1. Example FXO configuration:
 - a. The gateway will use 'One Stage' dialing to pass the digits received in the TO header of the INVITE from the IP side to the PSTN/PBX after detecting dial tone from the PSTN/PBX

constant Search				
Search	-			
Basic OFull	Dialing Mode	One Stage	~	2
Network Settings	Waiting for Dial Tone	Yes	~	2
Media Settings	Time to Wait before Dialing [msec]	1000		
Protocol Configuration	Ring Detection Timeout [sec]	8		
Advanced Applications	Reorder Tone Duration [sec]	255		
Voice Mail Settings	Answer Supervision	No	 Image: A set of the set of the	
Erx0 Settings	Rings before Detecting Caller ID	1	~	
	Send Metering Message to IP	No	~	
	Disconnect Call on Detection of Busy Tone	Enable	~	
	Disconnect On Dial Tone	Disable	~	
	Guard Time Between Calls	1		

- 10. Configure "Automatic Dialing" page:
 - a. By default, when a call is received from the PSTN/PBX the gateway will SEIZE the POTS line and play dial tone to the PSTN/PBX caller and waits for digits to be dialed. In some cases, the gateway is configured to automatically send the PSTN/PBX calls to a predetermined IP endpoint.
 - i. Example FXO configuration:
 - 1. FXO port # 1 will automatically dial '2001'
 - 2. FXO port # 2 will automatically dial '7049'
 - 3. FXO ports # 3-4 will provide dial tone to the PSTN/PBX side

Port	Destination Phone Number	Status
Port 1 FXS	2001	Enable 💌
Vetwork Settings Media Settings	7049	Enable 💌
Protocol Configuration Port 3 FXS		Enable 🗸
* Protocol Definition Port 4 FXS		Enable 🗸
Port 5 FXO		Enable 🗸
Profile Definitions Port 6 FXO		Enable 🗸
Port 7 FXO		Enable 🗸
Automatic Dialing Port 8 FXO		Enable V
Caller Display Information Call Forward Call Waiting Call Waiting Call Waiting Call Waiting Caller Upermissions Call Waiting Caller Upermissions Call Waiting Cal		

After configuring the AudioCodes MP-114 FXO port for T.38 faxing, the INI file should be similar to the following exported INI file.

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

;Board: MP-114 FXS_FXO ;Serial Number: 1239153 ;Slot Number: 1 ;Software Version: 5.60A.025.005 ;DSP Software Version: 204IM => 560.12 ;Board IP Address: 10.128.28.20 ;Board Subnet Mask: 255.255.252.0 ;Board Default Gateway: 10.128.28.1 ;Ram size: 32M Flash size: 8M ;Num of DSP Cores: 1 Num DSP Channels: 4 ;Profile: NONE

[SYSTEM Params]

SyslogServerIP = 10.15.6.100

[BSP Params]

[Analog Params]

FXSLoopCharacteristicsFilename = 'MP11x-02-1-FXS_16KHZ.dat'

[ControlProtocols Params]

[MGCP Params]

[MEGACO Params]

EP_Num_0 = 0 EP_Num_1 = 1 EP_Num_2 = 0 EP_Num_3 = 0 EP_Num_4 = 0

[Voice Engine Params]

FarEndDisconnectSilencePeriod = 60 CallProgressTonesFilename = 'usa_tones_12.dat' ECNLPMode = 1 BrokenConnectionEventTimeout = 3 CNGDetectorMode = 2 RFC2833PayloadType = 101 DTMFDetectorSensitivity = 1

[WEB Params]

[SIP Params]

MAXDIGITS = 4 ISTWOSTAGEDIAL = 0 ENABLECURRENTDISCONNECT = 1

```
ENABLEREVERSALPOLARITY = 1
CDRREPORTLEVEL = 1
GWDEBUGLEVEL = 5
ENABLEVOICEDETECTION = 1
DISCONNECTONBROKENCONNECTION = 0
MWIANALOGLAMP = 1
ENABLEMWI = 1
ISFAXUSED = 1
SUBSCRIPTIONMODE = 1
3XXBEHAVIOR = 1
[IPsec Params]
[SNMP Params]
 *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
 *** TABLE PREFIX ***
[PREFIX]
FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileId,
PREFIX_MeteringCode, PREFIX_DestPort, PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix, PREFIX_DestIPGroupID,
PREFIX_SrcHostPrefix, PREFIX_TransportType, PREFIX_SrcTrunkGroupID;
PREFIX 0 = 4444, 10.128.16.110, *, 0, 255, 0, -1, , -1, , -1, -1;
PREFIX 1 = 2222, 10.128.16.136, *, 0, 255, 0, -1, , -1, , -1, -1;
PREFIX 2 = 1000, 10.128.16.146, *, 0, 255, 0, -1, , -1, , -1, -1;
PREFIX 3 = *, 10.128.16.110, *, 0, 255, 0, -1, , -1, , -1;
[\PREFIX]
 *** TABLE CoderName ***
[CoderName]
FORMAT CoderName_Index = CoderName_Type, CoderName_PacketInterval, CoderName_rate, CoderName_PayloadType,
CoderName_Sce;
CoderName 0 = g711Ulaw64k, 20, 0, 255, 0;
CoderName 1 = g711Alaw64k, 20, 0, 255, 0;
[\CoderName]
 *** TABLE TrunkGroup ***
[TrunkGroup]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, 2001, 0, 255, 255;
TrunkGroup 1 = 1, 255, 2, 2, 2002, 0, 255, 255;
TrunkGroup 2 = 2, 255, 3, 3, 3001, 0, 255, 255;
TrunkGroup 3 = 2, 255, 4, 4, 1111, 0, 255, 255;
[\TrunkGroup]
```

18

```
*** TABLE PstnPrefix ***
[PstnPrefix]
FORMAT PstnPrefix Index = PstnPrefix DestPrefix, PstnPrefix TrunkGroupId, PstnPrefix SourcePrefix, PstnPrefix SourceAddress,
PstnPrefix_ProfileId, PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = 2001, 1, *, *, 0, -1, , ;
PstnPrefix 1 = 2002, 1, *, *, 0, -1, , ;
PstnPrefix 2 = 1111, 2, *, *, 0, -1, , ;
PstnPrefix 3 = *, 2, *, *, 0, -1, , ;
[\PstnPrefix]
  *** TABLE TxDtmfOption ***
[TxDtmfOption]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;
[ \TxDtmfOption ]
  *** TABLE TrunkGroupSettings ***
[TrunkGroupSettings]
FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId, TrunkGroupSettings_ChannelSelectMode,
TrunkGroupSettings_RegistrationMode, TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup;
TrunkGroupSettings 0 = 1, 0, 255, , , -1;
TrunkGroupSettings 1 = 2, 1, 255, , , -1;
[\TrunkGroupSettings]
  *** TABLE TargetOfChannel ***
[TargetOfChannel]
FORMAT TargetOfChannel_Index = TargetOfChannel_Destination, TargetOfChannel_Type;
TargetOfChannel 2 = 1111, 1;
TargetOfChannel 3 = 1111, 1;
[\TargetOfChannel]
  *** TABLE ProxySet ***
[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap;
ProxySet 0 = 0, 60, 0, 0;
```

```
[ \ProxySet ]
```

8. Frequently Asked Questions

- "I'm configured as near as possible to this the sample configuration described in this document, but calls are still not successful; what is my next step?"
 - → Provide this document to your gateway support.
 - → Ensure T.38 is enabled on the gateway.
 - → Confirm that basic network access is possible by pinging the gateway.
- "How do I obtain Wireshark traces?"
 - ➔ The traces can be viewed using the Wireshark network analyzer program, which can be freely downloaded from <u>http://www.wireshark.org</u>.
 - ➔ To view the call flow in Wireshark, open the desired network trace file and select "Statistics->VoIP Calls" from the drop down menu. Then highlight the call and click on the "Graph" button.