6.0A - AudioCodes MP-124D 24 port FXS MediaPack

1 Important Notes

- Check the SIP 3rd Party Validation Website for current validation status.The SIP 3rd Party Validation Website can be viewed at:

  http://testlab.inin.com or http://testlab.vonexus.com

- It is recommended to make basic changes to the I3 supplied .ini file then utilize the web interface for further configuration.
- To upgrade to version 6.0A, the device must first be upgraded to 5.8 or better. Upgrading 5.6 to 6.0 will cause the gateway to enter a fail state and the gateway will not boot. To recover from the failed state, the Audiocodes BootP program must be used to provide a firmware other than 6.0. Both 5.8 and 6.0 .cmp files have been provided.
- As this unit is a critical piece of the SIP infrastructure, it is highly recommended that DHCP not be used. A static IP address is the preferred method and is how the unit was configured during validation.
- The MinFlashhookTime needs to be set to 700 from 300 (default) and FlashhookPeriod needs to be set to 700 from 400 (default). These two parameters need to be set for flash hook to work properly. If the values were left at default values, pressing the flash hook button disconnects the connected call.
- When using SRTP the number of available channels decreases by ~25%.
- Homing redundancy behavior has been optimized. The gateway will now mark a proxy as "dead" when it fails to respond to an Invite or a keep-alive message. The gateway will then continue to monitor the status of the dead proxy and not send it invites until it responds to a keep-alive message. Previously, the gateway has been known to try each proxy in the proxy sets list even if a proxy has gone down.

2 Vendor Documentation

Documentation can be found on the CD shipped with the Media Packs.

3 Current Versions Verified

MP-124D Series: V6.00A.020.002

4 Install

Download the Media Pack files from the Interactive Intelligence Testlab website for the gateway:
http://testlab.inin.com or http://testlab.vonexus.com

Contained in the zip file will be the validated version of firmware (.cmp), as well as any supplemental configuration files, and a sample .ini file.

5 Configuration

Methods:
- Manipulation of the supplied .ini file, then uploading it via the web interface. This method is recommended for a quick start scenario. The I3 supplied .ini file, with just a few modifications will get a basic setup up and running. Please note, the I3 supplied .ini file is not meant to be a drop in configuration for every environment, it is provided as reference only.
- Web interface. (This method was used to configure the unit during validation) There are many advanced options that are exposed in the web interface. Caution should be exercised and the Audiocodes documentation should always be referenced when using the web interface configuration option.
- TFTP. This has not been tested.

Initial Setup:
- Unzip the ZIP file containing the MediaPack configuration files and firmware.
- Follow the AudioCodes instructions (on the CD or in the booklet shipped with the Mediant 2000 or in) for getting an IP address assigned to the MP box.
- Modify the I3 provided .ini file as appropriate.
- Start a web browser and type in the IP address of the MediaPack.
- The default user name is "Admin" (with a capital "A") and the password is "Admin" (with a capital "A"). It is recommended to change the password at this time for security reasons.
- "Terminate Analysis on Connect" under the "Session" page of the line configuration needs to be enabled for fax functionality.

Download Current Firmware (CMP file):
- From the main web screen, select *Software Update* from the icons on the left side of the page.
- Depending on what version of firmware is currently running on the MediaPack, two options will be available, "Software File-Download" or "Software Upgrade Wizard"
- Choose the appropriate option and load the validated version of firmware onto the MediaPack.
- To upgrade to version 6.0A, the device must first be upgraded to 5.8 or better. 5.6, specifically will cause the gateway to enter a fail state and the gateway will not boot. Both 5.8 and 6.0 .cmp files have been provided.
- After the validated firmware has been applied the MediaPack will restart. Once the restart is complete, return to the web interface and click on "Status & Diagnostics" on the left hand side of the screen. Then select "System Information" from the top of the screen. The displayed Version ID should match the firmware version as noted in section 1.1.3. If it does not, double check the upgrade steps and consult the Audiocodes documentation if necessary.

Download Auxiliary File(s):
- From the Software Update Section, select "Load Auxiliary Files"
- Click the browse button to select the auxiliary file to download, then click "Send File."
- During the validation procedure, the coefficient file and the call progress tones file were uploaded.
- Repeat the steps in this section any additional auxiliary files required for this installation.

Changing the Configuration:
- Prior to making any changes, the Audiocodes documentation should be consulted for information on configuration parameters, options and functions.
- Changes can be made via the web interface or the .ini file can be modified and re-downloaded to the MediaPack.
- Description of more significant .ini parameters. This is not a comprehensive list of all parameters found in the .ini file. A **bold** face parameter name indicates that it should be changed to represent specific site information.

FXS SIP Message Specifics	
Parameter	SIP Message Usage
[the dialed number]	Used in user portion in the SIP address in the To header.
ProxyName	Used in host portion in the SIP address in the To header.
CallerDisplayInfo*X*	Used in display name in the From header.
Channel2Phone or ChannelList	From MP: Used in user portion of the SIP address in the From header. To MP: Specifies the user portion of the SIP address needed to ring the phone.
SipGatewayName	Used in host portion in the SIP address in the From header.
FXO SIP Message Specifics	
Parameter	SIP Message Usage
TargetOfChannel*X*	Used in user portion in the SIP address in the To header.
ProxyName	Used in host portion in the SIP address in the To header.
CallerDisplayInfo*X*	Used in display name in the From header.
EnableCallerID	Values collected from the CO is used in user portion of the SIP address in the From header (if EnableCallerID=1).
Channel2Phone or ChannelList	Used in user portion of the SIP address in the From header (if EnableCallerID=0).
SipGatewayName	Used in host portion in the SIP address in the From header.

Fields common to both the MP-xxx FXS and FXO Units	
Parameter	Description
Channel Parameters	
IPPrecedence	Sets the 3 IP precedence layer 3 bits. *Values*: 0 through 7 *Example*: IPPrecedence = 5
IPTOS	Sets the 4 TOS layer 3 bits. *Values*: 0 through 15 *Example*: IPTOS = 0

DTMFTransportType	Specifies transport mechanism for DTMF tones
	Values:
	  0 (erase digits and do not relay)
	  1 (erase digits and relay to remote – proprietary)
	  2 (digits remain in voice stream – inband)
	  3 (erase digits and relay using RFC2833)
	Example: DTMFTransportType = 3

RTPDTMFRFC2833PayloadType	Payload number to use for RFC2833 packets. This value is only used when DTMFTransportType = 3. Values: 96-99, 105-127

CoderName	Which coders should be used. Each coder will be on its own line. Up to 5 can be given. The first one specified is given precedence.
	Values:
	  g711Ulaw64k,20
	  g711Alaw64k,20
	  g729
	  g7231
	  g726
	Note: This line can be specified multiple times.
	Example:
	  ;first coder used
	  CoderName = g711Ulaw64k,20
	  ; second coder used
	  CoderName = g7231

SIP Parameters	
MGControlProtocolType	Specifies call control protocol to use. 8 specifies SIP. *Note:* **This value should not be changed.**

	Value: 8, SIP *Example*: MGControlProtocolType = 8

IsProxyUsed	Using a Proxy (or Interaction Center Server) for all calls?
	Values:
	   0 (no, using the Prefix Routing Table)
	   1 (yes, the proxy is in the *ProxyIP* field)
	Example: IsProxyUsed = 1

ProxyIp	Proxy or Interaction Center Server IP address (if *IsProxyUsed=1*). *Note*: If a proxy is not used then the Prefix Routing Table is used. *Example*: ProxyIp = 172.16.129.160
IsRegisterNeeded	Whether to register. *Values*: 0 (no), 1 (yes) *Example*: IsRegisterNeeded = 0
ProxyName	Proxy Server Name. Used in host portion in the SIP address in the To header. *Example*: ProxyName = ICServer1.inin.com
SipGatewayName	Gateway Host Name. Used in host portion in the SIP address in the From header. Example: SipGatewayName = mp1xxfxo.inin.com

DNSPriServerIP
DNSSecServerIP
	Address of the primary and secondary DNS servers.
	Example:
	   DNSPriServerIP = 172.16.1.1
	   DNSSecServerIP = 172.16.1.2

Fields specific to the MP-xxx FXS Units	
Parameter	Description
IsDialNeeded	Is dialing required or should auto-dial (i.e. the bat phone) be used *Values*: 0 (no), 1 (yes) *Example*: IsDialNeed = 1
IsSpecialDigits	Is # needed as a dialed character? If set to 0 then # will terminate DTMF dialing. *Values*: 0 (no), 1 (yes) *Example*: IsSpecialDigits = 0
Endpoints	

Channel2Phone
or
ChannelList
	From MP FXS: Used in user portion of the SIP address in the From header.
	To MP FXS: Specifies the user portion of the SIP address needed to ring the phone.
	
	IMPORTANT: The values for these parameters must only be numbers. Dashes, letters, or any non-number character will give strange results.
	
	IMPORTANT: Use either Channel2Phone(s) or ChannelList (not both).
	
	Channel2Phone specifies the extension of a phone connected to a single port. One entry is required per active port/phone.
	Note: *Channel2Phone* can be specified multiple times.
	Values: Channel2Phone = 0 based port#, number (only umbers, no letters, no dashes)
	Examples:
	   Channel2Phone = 0,101
	   Channel2Phone = 1,116
	   Channel2Phone = 6,157
	
	ChannelList specifies a range of extensions beginning at a specified port, continuing for a specified number of ports and starting with a specified extension.
	
	Note: The *ChannelList* parameter could be used here

	instead of *Channel2phone* if the extensions are contiguous.
	Values: ChannelList = 0 based port#, number of ports, starting extension number
	Examples:
	  ChannelList = 0,8,201
	  ChannelList = 4,2,7500

Caller ID of each end point	

CallerDisplayInfo*X*	Used in display name in the From header. This line should only contain numbers and letters. *X* corresponds to the port number and is 0 based.
	Example:
	  CallerDisplayInfo0 = John Smith
	  CallerDisplayInfo1 = Jane Doe
	  CallerDisplayInfo2 = Ben Hamilton

Prefix Routing Table	

Prefix	Routing table based on dialed number. This is very useful for 911 dialing at remote sites ***Only*** used if *IsProxyUsed* = 0.
	Values: prefix, IP Address.
	Note: This line can be specified multiple times.
	Example (1.1.1.1 is a local gateway, 2.2.2.2 in the Interaction Center server):
	  Prefix = 911,1.1.1.1
	  Prefix = *,2.2.2.2
	Note: * is the only regular expression supported.

Fields specific the MP-xxx FXO Units	
Parameter	Description
General Parameters	
IsDialNeeded	Should dialtone be generated for PSTN to IP calls. *Values*: 0 (no), 1 (yes) *Example*: IsDialNeeded = 0
IsUseFreeChannel	Select the next free channel. *Values*: 0 (no), 1 (yes) *Example*: IsUseFreeChannel = 1
IsTwoStageDial	Should dialtone be generated for IP to PSTN calls. *Values*: 0 (no), 1 (yes) *Example*: IsTwoStageDial = 0
EnableCallerID	Enable CallerId *Values*: 0 (no), 1 (yes) *Example*: EnableCallerID = 1 If used, the caller id values collected will be used in user portion of the SIP address in the From header.
CurrentDisconnectDuration	Default: 900 Tested: 800 (for US) This parameter may need to be adjusted if the FXO MP fails to disconnect the analog line after the call ends.
Endpoints	
Channel2Phone **or** **ChannelList**	Used in user portion of the SIP address in the From header. **IMPORTANT: The values for these parameters must only be numbers. Dashes, letters, or any non-number character will give strange results.** **IMPORTANT: Use either Channel2Phone(s) or ChannelList (not both).** *Channel2Phone* specifies the line's phone number connected to a single port. One entry is required per active port/phone. *Note*: *Channel2Phone* can be specified multiple times. *Values*: Channel2Phone = 0 based port#, number (only umbers, no letters, no dashes) *Examples*: Channel2Phone = 0,101 Channel2Phone = 1,116 Channel2Phone = 6,157 *ChannelList* specifies a range of extensions beginning at a specified port, continuing for a specified number of ports and starting with a specified extension. *Note*: The *ChannelList* parameter could be used here instead of *Channel2phone* if the extensions are contiguous. *Values*: ChannelList = 0 based port#, number of ports, starting extension number *Examples*: ChannelList = 0,8,201 ChannelList = 4,2,7500
Caller ID of each end point	
CallerDisplayInfo*X*	Used in display name in the From header. This line should only contain numbers and letters. *X* corresponds to the port number and is 0 based. *Examples*: CallerDisplayInfo0 = Mary Smith CallerDisplayInfo1 = Acme Inc
Automatic dialed numbers	
TargetOfChannel*X*	Used in user portion in the SIP address in the To header. *X* corresponds to the port number and is 0 based. This will be the attribute Local_TN when the call arrives at the Interaction Center. *Examples*:

| | TargetOfChannel0 = 7151111 |
	TargetOfChannel1 = 7151112
TargetOfChannelX	Used in user portion in the SIP address in the To header. X corresponds to the port number and is 0 based. This will be the attribute Local_TN when the call arrives at the Interaction Center. *Examples*: TargetOfChannel0 = 7151111 TargetOfChannel1 = 7151112

Download Configuration file (INI file):

- From the MP main web screen, select *Software Update* from the icons on the left side of the page. The default user name is "Admin" (with a capital "A") and the password is "Admin" (with a capital "A").
- Choose *HTTP-download* from the icons on the top row of the page.
- In the "Send INI File" section, select the Browse button and locate the customized INI created above.   Press Send File and then, after the download is complete, Reset the MP box using the Reset option.

6 Security

To enable TLS the Media pack needs HTTPSCipherString set to All.
To do this:

Navigate to http://<Gateway IP address>/AdminPage
Select ini Parameters on the left
In the Parameter Name field, enter: HTTPSCipherString
In the Value field, enter: ALL
Select: Apply New Value
Select: Back to Main
Select: Maintenance
On the Maintenance page, select: BURN
On the Maintenance page, select: Reset

For the rest of the TLS/SRTP configuration refer to either the 5.2 User manual or the Interactive Intelligence document:

Securing AudioCodes Mediant and MediaPack Gateways with xIC

7 Redundant Proxy Configuration

The following configuration options are found under Protocol Management/Protocol Definition/Proxy & Registration in the web interface. In order for the Media Pack to function with redundant proxies, the following value must be set:

Parameter	Description
Proxy IP Address	IP Address of the primary proxy.
First Redundant Proxy IP Address	IP Address of the secondary proxy.
Redundancy Mode	Values: Homing, Parking Homing: Gateway will ALWAYS try the primary proxy on every call attempt. Parking: Gateway will "park" on the last known good proxy until that proxy fails.
Enable Proxy Keep Alive	Value: Using Options Gateway will sent OPTIONS messages to verify a proxy is online.
Enable Proxy Hot-Swap	Value: Enabled Gateway will attempt proxy failover if this is not set.

The following parameters are optional and enable advanced failover capabilities:

Parameter	Description
Enable Fallback to Routing Table	If none of the configured proxies are available, the gateway will attempt to deliver the call based upon its internal routing Tel to IP routing table.
Proxy Keep Alive Time	How often (in seconds) the OPTIONS messagew will be sent to verify a proxy is online.
Number of RTX Before Hot-Swap	How many re-transmits before failover takes place.