

AudioCode & Mitel

Analogue/Fax Integration Guide

Equipment	Model	Version
Mitel MCD	MXe III	7.0
AudioCode	MediaPack 11X	MP 11x

MCD

Licenses

- Make sure there is enough licenses. You will need the following licenses:
 - **IP User** – for SIP Generic Phones
 - **Compression license** for E2T compression
 - **T.38** codec licenses for T.38 Fax
 - **SIP trunks** for SIP link between MCD and ATA
 - **Analog licenses** for ONS port (if using ASU)

Licensed Options		
	Consumed	Allocated
Users		
IP Users	1	1
External Hot Desk Users	0	0
ACD Active Agents	0	0
HTML Applications	0	0
Analog Lines	22	22
MyVoice Business Console Active Operators	0	0
Multi-device Users	0	0
Multi-device Suites	0	0
Messaging		
Embedded Voice Mail	0	0
Embedded Voice Mail PMS	0	0
Trunking/Networking		
Digital Links	1	1
Compression		0
FAX Over IP (T.38)		8
SIP Trunks	5	120
Others		
IDS Connection	0	0
MLPP	0	0

Class of Service

- Many different options may be required for your site deployment, but ensure that “Public Network Access via DPNSS” Class of Service Option is configured for all devices that make outgoing calls through the SIP trunks in the MCD
 - **Public Network Access via DPNSS** set to Yes

- **Campon Tone Security/FAX Machine** set to Yes
- **Busy Override Security** set to Yes

Trunk Attributes

- **Class of Service** should match that one programed above.

Trunk Attributes	
Trunk Service Number	6
Release Link Trunk	<input type="button" value="No"/>
Call Recognition Service	Off
Direct Inward Dialing Service	<input checked="" type="radio"/> Off <input type="radio"/> On
Class of Service	14
Class of Restriction	5
Baud Rate	300
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Label	ATA T38

Network Element

- Create a network element for each MP 11X gateway.
 - **Network Element Type** - Other
 - **FQDN or IP Address** should match the IP address of the MP 11X
 - Use port 5060

Network Elements	
Name	AudioCode
Type	Other
FQDN or IP Address	10.64.32.50
Local	False
Version	
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	default
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

SIP Peer Profile

- Create a SIP Peer Profile with the following attributes:
 - **Network Element:** The selected SIP Peer Profile needs to be associated with previously created “AC-MP-118” Network Element.
 - **Address Type:** Select IP address.
 - **Outbound Proxy Server:** Select the Network Element previously configured for the Outbound Proxy Server.

- **Calling Line ID:** The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the DID ranges for CPN Substitution form (see DID Ranges for CPN Substitution). Then select the appropriate numbers in the Outgoing DID Ranges in this form (SIP Peer Profile).
- **Trunk Service Assignment:** Enter the trunk service assignment previously configured.
- **SMDR:** If Call Detail Records are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank).
- **Maximum Simultaneous Calls:** This entry should be configured to maximum number of SIP trunks provided by MP 11X

Basic

<u>SIP Peer Profile Label</u>	AudioCode
<u>Network Element</u>	AudioCode
Local Account Information	
<u>Registration User Name</u>	
<u>Address Type</u>	IP Address: 10.64.32.248
Administration Options	
<u>Interconnect Restriction</u>	1
<u>Maximum Simultaneous Calls</u>	5
<u>Minimum Reserved Call Licenses</u>	5
Administration Options	
<u>Outbound Proxy Server</u>	
<u>SMDR Tag</u>	0
<u>Trunk Service</u>	6
<u>Zone</u>	1
<u>User Name</u>	
<u>Password</u>	*****
<u>Confirm Password</u>	*****
<u>Authentication Option for Incoming Calls</u>	No Authentication
<u>Subscription User Name</u>	
<u>Subscription Password</u>	*****

Call Routing

<u>Alternate Destination Domain Enabled</u>	No
<u>Alternate Destination Domain FQDN or IP Address</u>	
<u>Enable Special Re-invite Collision Handling</u>	No
<u>Only Allow Outgoing Calls</u>	No
<u>Private SIP Trunk</u>	No
<u>Reject Incoming Anonymous Calls</u>	No
<u>Route Call Using P-Called-Party-ID (if present)</u>	No
<u>Route Call Using To Header</u>	No

Calling Line ID

Default CPN	
Default CPN Name	
CPN Restriction	No
Public Calling Party Number Passthrough	No
Strip PNI	No
Use Diverting Party Number as Calling Party Number	No
Use Original Calling Party Number If Available	No

SDP Options

Allow Peer To Use Multiple Active M-Lines	Yes
Allow Using UPDATE For Early Media Renegotiation	Yes
Avoid Signaling Hold to the Peer	Yes
AVP Only Peer	Yes
Enable Mitel Proprietary SDP	No
Force sending SDP in initial Invite message	No
Force sending SDP in initial Invite - Early Answer	No
Ignore SDP Answers in Provisional Responses	No
Limit to one Offer/Answer per INVITE	Yes
NAT Keepalive	Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
Renegotiate SDP To Enforce Symmetric Codec	No
Repeat SDP Answer If Duplicate Offer Is Received	No
Restrict Audio Codec	No Restriction
RTP Packetization Rate Override	Yes
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	No
Suppress Use of SDP Inactive Media Streams	Yes

Signalling and Header manipulation

Trunk Group Label	
Allow Display Update	No
Build Contact Using Request URI Address	No
De-register Using Contact Address not *	Yes
Disable Reliable Provisional Responses	Yes
Disable Use of User-Agent and Server Headers	No
E.164: Enable sending '+'	No
E.164: Add '+' if digit length > N digits	0
E.164: Do not add '+' to Emergency Called Party	No
E.164: Do not add '+' to Called Party	No
Force Max-Forward: 70 on Outgoing Calls	No
If TLS use 'sips:' Scheme	No
Ignore Incoming Loose Routing Indication	No
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Require Reliable Provisional Responses on Outgoing Calls	Yes
Signal Privacy (if enabled) on Emergency Calls	No
Suppress Redirection Headers	No
Use Fixed Retry Time for 491	No
Use Privacy: none	No
Use P-Asserted Identity Header	Yes
Use P-Asserted Identity for Billing	No

Use P-Call-Leg-ID Header	No
Use P-Preferred Identity Header	No
Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No
Use user=phone	No

Timers

Keep-Alive (OPTIONS) Period	120
Registration Period	300
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	90
Session Timer: Local as Refresher	No
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

User and Devices Configuration

- Create Generic SIP Phones for the number of Analog Ports needed on the MP 11X ATA.
 - Be sure to give the SIP devices the appropriate COS and COR.

User Profile

Number	<input type="text" value="30980"/>
Service Label	<input type="text" value="Phone Service"/>
Directory Name	<input type="text" value="Audio Code"/>
Prime Name	<input checked="" type="radio"/> No <input type="radio"/> Yes
Privacy	<input checked="" type="radio"/> No <input type="radio"/> Yes
Hot Desking User	<input checked="" type="radio"/> No <input type="radio"/> Yes
Device Type	<input type="text" value="Generic SIP Phone"/>
Service Level	<input type="text" value="Full"/>
Home Element	UCISCV248
Secondary Element	<input type="text" value="Not Assigned"/>
Local-only DN	<input type="checkbox"/>
ACD Enabled	<input type="checkbox"/>

Access and Authentication

- In the tab below, add the extension number as the SIP Password.

User PIN	<input type="password" value="••••••••"/>
Confirm User PIN	<input type="password" value="••••••••"/>
SIP Password	<input type="password" value="••••••••"/>
Confirm SIP Password	<input type="password" value="••••~•••"/>
Wireless PIN	<input type="password" value=""/>
Confirm Wireless PIN	<input type="password" value=""/>

Keys

- Create a Multicall Key and set the Directory as itself with Ring as the Ring Type.

	Button Number	Label	Line Type	URL	Button Directory Number	Ring Type	MiXML Application Feature	Phone Application Feature
▶	2	ATA	Multicall		30980	Ring	Not Assigned	
▶	3		Not Assigned				Not Assigned	

Automatic Route Selection

- Create a Route for the SIP peer profile
- In the ARS digit table, add a leading outgoing digit for this route followed by the extension number.
- Use the appropriate Digit Modification for the route.
- Use appropriate Class of Restrictions


MP 11X

VoIP


- Connect to default IP – 10.1.10.10
 - Username – Admin
 - Password – Admin
- >> Network Setting >> IP Interface Table

▼ Single IP Settings	
IP Address	10.64.32.50
Subnet Mask	255.255.255.0
Default Gateway Address	10.64.32.1

- >> SIP Definitions >> Proxy & Registration

Use Default Proxy	Yes ▼
Proxy Set Table	
Proxy Name	<input type="text"/>
Redundancy Mode	Parking ▼
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable ▼
Prefer Routing Table	No ▼
Use Routing Table for Host Names and Profiles	Disable ▼
Always Use Proxy	Enable ▼
Enable Registration	Enable ▼
Registrar Name	<input type="text"/>
Registrar IP Address	<input type="text"/>
Registrar Transport Type	Not Configured ▼
Registration Time	300
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable ▼
ReRegister On Connection Failure	Disable ▼
Gateway Name	<input type="text"/>
Gateway Registration Name	<input type="text"/>

Subscription Mode	Per Endpoint ▼
User Name	
Password	Default_Passwd
Cnonce	Default_Cnonce
Registration Mode	Per Endpoint ▼

- Select  arrow below the Proxy Set Table
 - Add in the IP address of the MCD
 - Transport type – UDP

	Proxy Address	Transport Type
1	10.64.32.248	UDP ▼
2		▼
3		▼
4		▼
5		▼

Enable Proxy Keep Alive	Disable ▼
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable ▼
Is Proxy Hot Swap	No ▼

- >> Coders and Profiles >> Coders – add in the codec you need to use.

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law ▼	20 ▼	64 ▼	8	Disabled ▼
T.38 ▼	N/A ▼	N/A ▼	N/A	N/A ▼
▼	▼	▼		▼
▼	▼	▼		▼

- >>GW and IP to IP >> DTMF and Supplementary >> DTMF & Dialling

Max Digits In Phone Num	5
Inter Digit Timeout [sec]	4
Declare RFC 2833 in SDP	Yes ▼
1st Tx DTMF Option	RFC 2833 ▼
2nd Tx DTMF Option	▼
RFC 2833 Payload Type	101
Default Destination Number	1000

- >>SIP Definitions >> Advanced Parameters – change the following

▼ Disconnect and Answer Supervision		
Polarity Reversal	Enable	▼
Current Disconnect	Enable	▼

- >>GW and IP to IP >> Routing >> IP to Hunt Group Routing

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	Source IP Group ID
1	*	*	*		1	-1
2						

- >>GW and IP to IP >> Hunt Group >> Endpoint Phone Number
 - Enter the Generic Sip extension number here

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	30980	1	0
2				

- >>GW and IP to IP >> Hunt Group >> Hunt Group Settings

	Hunt Group ID	Channel Select Mode	Registration Mode
1	1	By Dest Phone Number ▼	Per Endpoint ▼
2		▼	▼

- >>GW and IP to IP >> Analog Gateway >> Authentication
 - Use the password programmed in the Generic SIP device of the MCD

Gateway Port	User Name	Password
Port 1 FXS	30980	*****
Port 2 FXS		

- >>GW and IP to IP >> Analog Gateway >> Caller Display Information
 - Give an extension name to this port

Gateway Port	Caller ID/Name	Presentation
Port 1 FXS	30980FX	Allowed ▼
Port 2 FXS		Allowed ▼
Port 3 FXS		Allowed ▼
Port 4 FXS		Allowed ▼

- >>GW and IP to IP >> Analog Gateway >> Caller ID Permissions

Gateway Port	Caller ID
Port 1 FXS	Enable ▼
Port 2 FXS	▼
Port 3 FXS	▼
Port 4 FXS	▼