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PBX Configuration

Avaya ACM/Definity

Enable Adjunct Supervision for each analog port connected to the DMG.

Mitel

Enable System Option 22 (Last Party Clear: Dial Tone) for all analog Mitel ports connected to the DMG.

DMG 1000 Configuration

Initial Setup

1. You must first add the DMG to your network. By default the DMG is configured for ip address 10.12.13.74.
2. Default username is **admin** and the default password is **IpodAdmin**. The password is case-sensitive.

The sections below correspond with the DMG menu items on the left of the configuration. Here you can see the default screen your presented with upon login.

The screenshot shows the Dialogic web interface for the DMG 1000. The top navigation bar includes the Dialogic logo, a breadcrumb trail 'Status > Summary', and a 'Ports' section with seven green indicator lights. The left sidebar contains a menu with categories: Status, Configuration, Diagnostics, and System. The main content area displays two tables:

General Status	
MAC	
IP	
Start Time	
Up Time	
Device Status	
SNMP System Name	
SNMP System Location	
SNMP System Contact	

Version Information	
Description	Version
Gateway Application (ROM)	
Gateway Application	
Main Board Boot (ROM)	
DSP Firmware (ROM)	
DSP Firmware	
Telephony Interface Application	
Telephony Interface Firmware	
Telephony Interface Boot	
Telephony Interface ID	
Adept Config	

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IP

IP Settings, LAN1	
MAC	00-a0-e6-89-d5-7c
* Client IP Address	10.12.13.74
* Client Subnet Mask	255.255.255.0
* Default Network Gateway Address	0.0.0.0
* BOOTP Enabled	No ▼
* SNTP Server IP Address	

Client IP Address

Set the IP settings to match the network.

Mgmt Protocols

All fields should remain as default shown here.

Management Protocols	
E-mail	
E-Mail Alarms Enabled	No ▼
E-Mail Minimum Alarm Severity	Info ▼
Destination E-Mail List	
E-Mail Server IP Address	
Source E-Mail Address	alarm@pbxgw.com
SysLog	
* SysLog Server IP Address	
Alarms to Syslog Enabled	No ▼
SysLog Minimum Alarm Severity	Info ▼
Diagnostics Trace to SysLog Enabled	No ▼
SNMP	
SNMP Traps Enabled?	No ▼
SNMP Minimum Alarm Severity	Info ▼
SNMP Trap IP List	255.255.255.255
* SNMP Community Name	public
* SNMP System Name	
* SNMP System Contact	
* SNMP System Location	
Web Server	
* HTTP Server Enabled	Yes ▼
* HTTPS Server Enabled	No ▼
Telnet	
* Telnet Server Enabled	Yes ▼
Serial Ports	
* Maintenance Port Enabled	Yes ▼

Routing Table

All fields should remain as default shown here.

Router Configuration

Inbound TDM Rules
 Inbound VoIP Rules
 TDM Trunk Groups
 VoIP Host Groups

Inbound TDM Rules				
Select	Enable	Rule Label	Request Type	Trunk Group
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	InboundTdm	Any	TdmAll

1

Move Selected Row:

Maximum Number of Inbound TDM Rules: 40

Detailed Configuration for Inbound TDM Rule: **InboundTdm**

Inbound TDM Request Matching					
<input type="button" value="Hide"/> CPID Matching					
Calling Party		Called Party		Redirecting Party	
Number	*	Number	*	Number	*
Name	*	Name	*	Name	*

Outbound Routes					
<input type="button" value="Hide"/> Device Selection					
Outbound Destination	VoIP	Host Group	VoipGroup-1	Route Method	Bridged

<input type="button" value="Hide"/> CPID Manipulation					
Calling Party		Called Party		Redirecting Party	
Number	S	Number	D	Number	R
Name	S	Name	D	Name	R

Select Primary / Alternate Route

Primary
 Alt-1
 Alt-2
 Alt-3
 Alt-4

Routing Table > VoIP Host Groups

In order to route calls to the DuVoice system the IP address must be entered under VoIP Host Groups.

Router Configuration

Inbound TDM Rules
 Inbound VoIP Rules
 TDM Trunk Groups
 VoIP Host Groups

Select	Name	Load-Balanced	Fault-Tolerant	Network Group
1	VoipGroup-1	false	false	Network Group #1

Maximum Number of VoIP Host Groups: 10

The selected Host Group is referenced by the following rules:

[inbound TDM] InboundTdm (Primary Route)

Host List

VoipGroup-1	Delete
	Delete

VoipGroup-1

Enter the IP address of the DuVoice system.

TDM > Analog

The DMG can be configured to integrate in two ways. The first is for the DMG to collect and interpret the integration then route the call to DuVoice system, the other is for the DMG to route the call to the DuVoice and once connected interpret the integration. The integration will operate best using option 1. If you find the integration is not operating correctly try option 2.

Option 1 - Preferred Method

TDM Analog Settings	
Timing	
Flash Hook (ms)	800
Wait for Dial Tone after Flash Hook	Yes ▼
Delay After Flash-Hook (ms)	2000
Loop Current Off Debounce (ms)	5000
Incoming Rings Before Answer	1
Ringing Timeout (ms)	6000
Feature Code	
Transfer Feature Code	!
Consult Call Dial Tone Drop Code	!
Consult Call Proceeding Drop Code	!!
Consult Call Busy Drop Code	!!
Consult Call Error Drop Code	!!
Consult Call Connected Drop Code	!!
Consult Call Disconnected Drop Code	!!
Message Waiting Control	
MWI Confirmation Tone	No ▼
Use Same Port for Mwi Clear/Set	Yes ▼
CPID Settings	
Analog Interface Type	PBX ▼
Central Office CID Type	None ▼
Central Office CID Alert Type	Pause in Ring Cycle ▼
FSK CID expiration	10000
FSK CID timeout	5000
Auto-Answer Inbound TDM Calls (Type II CPID)	Yes ▼
Initial Wait for Inband CPID (ms)	2000
Inband CPID Complete Timeout (ms)	300
CID to First Ring Timeout (ms)	2000
Rx/Tx Gain Control	
Analog Receive Gain (dB)	0 ▼

Loop Current Off Debounce(ms)

Set to 5000.

Auto-Answer Inbound TDM Calls (Type II CPID)

Set to Yes.

TDM > General

TDM General Settings	
* PCM Coding	uLaw ▼
Minimum Call Party Delay (ms)	2000
Maximum Call Party Delay (ms)	5000
Dial Digit On Time (ms)	100
Dial Inter-Digit Time (ms)	100
Dial Pause Time (ms)	2000
Turn MWI On FAC	
Turn MWI Off FAC	
Outbound Call Connect Timeout (ms)	10000
Wait for Ringback/Connect on Blind Transfer	Yes ▼
* Hunt Group Extension	
Disconnect on Fax Cleardown Tone	No ▼
Connect Outbound Call On DTMF	No ▼

Minimum Call Party Delay (ms)

Set to 2000.

Maximum call Party Delay (ms)

Set to 5000. You may need to increase this value depending on your integration. The value should be set as low as you can to reduce delays.

Option 2

TDM Analog Settings	
Timing	
Flash Hook (ms)	800
Wait for Dial Tone after Flash Hook	Yes ▼
Delay After Flash-Hook (ms)	2000
Loop Current Off Debounce (ms)	5000
Incoming Rings Before Answer	1
Ringing Timeout (ms)	6000
Feature Code	
Transfer Feature Code	!
Consult Call Dial Tone Drop Code	!
Consult Call Proceeding Drop Code	!!
Consult Call Busy Drop Code	!!
Consult Call Error Drop Code	!!
Consult Call Connected Drop Code	!!
Consult Call Disconnected Drop Code	!!
Message Waiting Control	
MWI Confirmation Tone	No ▼
Use Same Port for Mwi Clear/Set	Yes ▼
CPID Settings	
Analog Interface Type	PBX ▼
Central Office CID Type	None ▼
Central Office CID Alert Type	Pause in Ring Cycle ▼
FSK CID expiration	10000
FSK CID timeout	5000
Auto-Answer Inbound TDM Calls (Type II CPID)	No ▼
Initial Wait for Inband CPID (ms)	2000
Inband CPID Complete Timeout (ms)	300
CID to First Ring Timeout (ms)	2000
Rx/Tx Gain Control	
Analog Receive Gain (dB)	0 ▼

Loop Current Off Debounce(ms)

Set to 5000.

Auto-Answer Inbound TDM Calls (Type II CPID)

Set to No.

TDM > General

TDM General Settings	
* PCM Coding	uLaw
Minimum Call Party Delay (ms)	0
Maximum Call Party Delay (ms)	0
Dial Digit On Time (ms)	100
Dial Inter-Digit Time (ms)	100
Dial Pause Time (ms)	2000
Turn MWI On FAC	
Turn MWI Off FAC	
Outbound Call Connect Timeout (ms)	10000
Wait for Ringback/Connect on Blind Transfer	Yes
* Hunt Group Extension	
Disconnect on Fax Cleardown Tone	No
Connect Outbound Call On DTMF	No

Minimum Call Party Delay (ms)

Set to 0.

Maximum call Party Delay (ms)

Set to 0.

MWI Settings

Use these settings for either option one or two.

Turn MWI On FAC and Turn MWI Off FAC

Avaya Definity

- Set MWI values to:
 - TURN MWI On FAC to *4
 - TURN MWI Off FAC to #4

Mitel

- Set MWI values to:
 - TURN MWI On FAC to #21
 - TURN MWI Off FAC to #22
 - Enable COS Option 259 (Message Sending).
 - Enable COS Option 216 (Data Security).

Other PBX's

Set the codes to match your PBX settings.

- You must set your PBX to send a dial tone at disconnect or loop current drop.
 - Mitel dial tone** - Enable System Option 22 (Last Party Clear: Dial Tone) for all analog Mitel ports connected to the DMG
 - Avaya ACM/Definity** enable Adjunct Supervision for each analog port for the DMG

TDM > CPID Parsing

These settings are for Avaya Definity. These values are from the configuration file available by Dialogic distributed in the firmware upgrade file. This is typically blank and must be configured on a per PBX basis. Prebuilt CPID files are available for import from <http://www.duvoice.com/downloads>.

```
rule #00#\d(3-10)##  
src_number 1  
reason direct
```

```
rule #00#\d(3-10)#\d(3-10)#  
src_number 1  
dst_number 2  
reason no-answer
```

```
rule #02#\d(3-10)#\d(3-10)#  
src_number 1  
dst_number 2  
reason no-answer
```

```
rule #01#\d(1-2)##  
reason trunk
```

```
rule #01#\d(1-2)  
reason opening
```

```
rule #03##\d(3-10)#  
dst_number 1  
reason no-answer
```

```
rule #04##\d(1-2)#  
reason trunk
```

VOIP > General

All fields should remain as default shown here.

Voip General Settings	
User-Agent	
* Host and Domain Name	pbxgw.default.com
User-Agent Header Value	PBX-IP Media Gateway
Call as Domain Name?	No ▼
Invite Expiration (sec)	120
Reliable Provisional Responses	Supported ▼
Server	
* DNS Server Address	
* DNS Server Address (Secondary)	
DNS Translation of Phone Numbers	No ▼
TCP/UDP	
* UDP/TCP Transports Enabled	Yes ▼
* TCP/UDP Server Port	5060
TCP Inactivity Timer (sec)	90
TLS	
* TLS Transport Enabled	No ▼
* TLS Server Port	5061
* SSL TLS Protocol	SSLv3_TLsv1 ▼
* Mutual TLS Authentication Required	Yes ▼
TLS Inactivity Timer (sec)	30
Verify TLS Peer Certificate Date	Yes ▼
Verify TLS Peer Certificate Trust	Yes ▼
Verify TLS Peer Certificate Purpose	Yes ▼
Timing	
T1 Time (ms)	500
T2 Time (ms)	4000
T4 Time (ms)	5000
* T1 Multiplier	64
Monitoring	
Monitor Call Connections	No ▼
Call Monitor Interval (sec)	60
* VoIP Host Monitor Interval (sec)	30
* Proactive DNS Monitoring	No ▼
QoS	
* Call Control QoS Byte	0

VoIP > Network Groups

All fields should remain as default shown here.

VoIP Network Group Configuration	
Network Group	
Network Group Label	Network Group #1
Transport	
Transport Protocol	UDP
SIPS URI Scheme	No
URI Parameters	
User Phone Parameter	Yes
Local Phone Context	
Remote Phone Context	
Diversion Header Format	TEL
Proxy	
Primary Proxy Server Address	
Primary Proxy Server Port	5060
Backup Proxy Server Address	
Backup Proxy Server Port	5060
Proxy Query Interval (sec)	30
Registration	
Registration Server Address	
Registration Server Port	5060
Registered User	
Gateway Name	
Registration Expiration (sec)	120
Audio	
Codec #1	G.711u
Codec #2	G.711a
Codec #3	None
Low Bit Rate Codec	G.723.1 [Modify]
Packet Time (ms)	30
SRTP	
SRTP Preference	RTP_Only
Authentication Tag Length	80
MKI on Transmit Stream	Yes
Key Derivation Enable	No
Key Derivation Rate	16
Window Size Hint	64
UnEncrypted SRTP Enable	No
UnEncrypted SRTCP Enable	No
UnAuthenticated SRTP Enable	No

VoIP > Media

VoIP Media Settings	
Early Media	
RFC 3960 Early Media Support	OnDemand ▼
Send Early 183 Progress Response	No ▼
Send Early 180 Ringing Response	Yes ▼
Require Reliable Provisional Responses	No ▼
Audio	
* Low Bit Rate Codec	G.723.1 ▼
Signaling Digit Relay Mode	Off ▼
Voice Activity Detection	Off ▼
Continue Ringback on CN	Yes ▼
Acceptable Media	RTP_S RTP ▼
Packet Time (ms) for Inbound VoIP	30 ▼
Digit Relay Mode	RFC2833 ▼
Telephone-Event Payload Type	101
Fax	
Fax IP-Transport Mode	T.38 ▼
Fax Server Host	
Fax Server Network Group	▼
Fax/Modem Tone Relay Mode	RFC2833 ▼
RTP	
* RTP Start Port	49000
* RTP End Port	50000
* RTP Source IP Address Validation	On ▼
* RTP Source UDP Port Validation	On ▼
RTP QoS Byte	0

Voice Activity Detection

Set to off. Setting this value to off will reduce clipping during audio recordings on calls with low db levels.

VoIP > Authentication

All fields should remain as default shown here.

VoIP Authentication		
<input checked="" type="radio"/> Inbound VoIP [Server] <input type="radio"/> Outbound VoIP [Client]		
Inbound VoIP Configuration		
Inbound Authentication Enabled	No	
Gateway Realm	default.gw.com	
Algorithm	MD5	
Methods to Challenge		
<input type="checkbox"/> Invite <input type="checkbox"/> Register <input type="checkbox"/> Notify <input type="checkbox"/> Info <input type="checkbox"/> Bye <input type="checkbox"/> Refer <input type="checkbox"/> Options		
Users		
Realm	User Name	Password
<input type="button" value="Add Entry"/>		

DuVoice Configuration

Lines

SIP Line 1
X

Port Configuration | SIP Configuration | SIP

PBX Port Integration

Extension number:

Hunt group extension is a member of:

PBX integration file:

Port Owner / Location Information

External IVR filename:

Assigned location:

Application:

Owner mailbox number:

Extension number

Set the extension number for each line to a matching number associated with an extension located on the DMG.

Hunt group extension is a member of

Set the hunt group field to the hunt group number assigned to the DMG extensions.

SIP Line 1

Port Configuration | SIP Configuration | SIP

Display name: Account name:

User agent: Password:

Local Port: Realm:

Optional endpoint override:

Enable Register

OK Cancel Apply Help

Enable Register

Do not check the Register field for any line.

SIP Configuration

Connectors

- System Details
 - Cisco
 - Inventory Server
 - IP Office
 - LDAP
 - PMS Pass-through
 - Room Status Server
 - ShoreTel
 - SIEMENS
 - SIP**
 - Univerge / Sphere

SIP

Location:

Registrar address: Port:

Dialogic IP Address:

Register expire time: seconds

SIP Trunk

OK Cancel Apply

Registrar address

Set the registrar address to the IP address of the DMG.

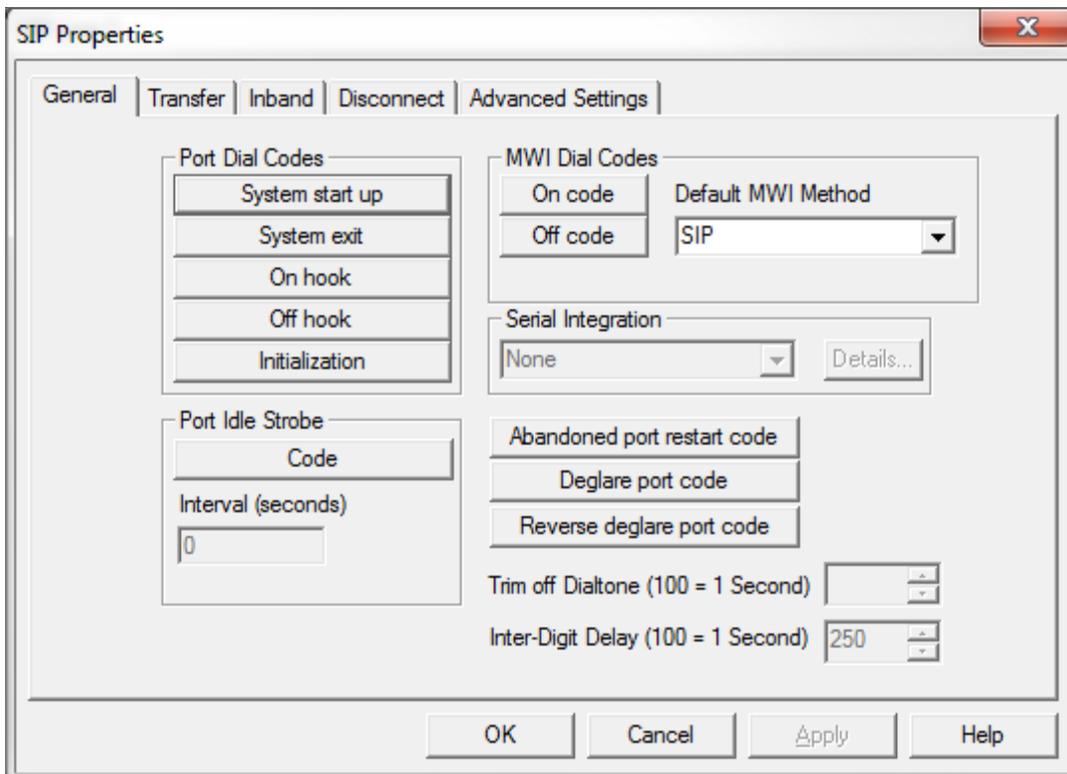
Registrar expire time

Set the Register expire time to what the DMG is configured for in VoIP General under the field Invite Expiration (sec). The default is 120.

MWI

System Configuration

Confirm the setting for Default MWI Method for the active template is set to SIP.



Mailbox Administration

Confirm the MWI On and MWI Off templates are configured as follows. MWI On shown here as an example.

MWI On

Notification

Definition

Event: all messages

Address: MWI 0

Technique: Message Waiting Indicator Off

Light for every message

Method: SIP

Initial Delay: 0 minutes.

Retry Interval: 1 minutes.

Do not exceed: 5 attempts.

Schedule

Days of the week this template is active:

Su M Tu W Th F Sa

Time period during which this notification is active:

Starting at: 12:00 AM

Ending at: 12:00 AM

OK Cancel

Method

Set to SIP for both MWI On and MWI Off templates. You may need this set to SIP+PMS if MWI should be sent to the PMS as well as the PBX.

Do not exceed

Set to 5 for both MWI On and MWI Off templates.