

Title	Digital Media Gateway using Digital
Document	dmg1000-digital-sip-in
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DuVoice Versions	5.XX
Switch Versions	N/A

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DMG 1000 Configuration

IP (default)

Dialogic DMG's are shipped from the factory with the IP address 10.12.13.74. This IP Address will need to be changed to match the existing network configuration. This can be done using either a small router configured temporally for this sub-net or by using a cross-over cable with the a system configured for the same sub-net.

Default username is **admin** and password **IpodAdmin** (case-sensitive).

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			

Mgmt Protocols (default)

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 (default)

Failure to enter the routing IP address will result in calls from the PBX not being directed to the DuVoice system.

			Ro	outer Configuration	n		
Inbound TDM Rul	es 🔘 Inbound VoIP Rules	🔘 та	OM Trunk G	roups 🔘 VoIP Hos	t Groups		
Select Enable	Rule Labe	1	1	Request T	ype	Tr	unk Group
	InboundTdm			Any		TdmAll	^
	Jineeanaren			1.49			
							+
	Move	e Sele	cted Row:	Up Down	To Position	1	
		_				Maximum	Number of Inbound TDM Rules: 40
Add Rule Delete R	lule						
Detailed Configuration	for Inbound TDM Rule: Inboun	dTdm					
			Tabawa		tation -		
Hide CPID Mat	china		Inboun	d IDM Request Ma	tening		
(alling Party			Called Party			Redirecting Party
Number *		Num	nber	*		Number	*
Name *		Nam	ne	*		Name	•
				-	1		·]
				Outbound Routes			
Device Se	election						
Outbound	VolP	-	Host	VoipGroup-1	•	Route	Bridged 🗸
Hide CPID Mar	ipulation		Group	,		Method	
Calling Party Called Party Redirecting Party							
Number	S		Number	D		Number	R
Name	s		Name	D		Name	R
Hax Select Primary / Alternate Route							
Drimper Alt=1 Alt=2 Alt=2 Alt=4 Add Alternate Route							
	te Delete Delete						
Dele		Delet	=				

Routing Table > VoIP Host Groups

Enter the IP address of the DuVoice system under VoipGroup-1

	Router Configura	tion					
🔘 Inbound TDM Rules 🔘 Inbound VoIP Rules 🔘 TDM Trunk Groups 🔘 VoIP Host Groups							
Select Name	Load-Balanced	Fault-Tolerant	Network Group				
VoipGroup-1	false 💌	false 💌	Network Group #1				
	,						
			*				
Maximum Number of VoIP Host Groups: 10							
Add Host Group Delete Host Group							
The selected Host Group is referenced by the following rul	es:		Host List				
[inbound TDM] InboundTdm (Primary Route)		VoipGrou					
			Delete				
			Add Host				
	.::						

TDM > Digital

Choose the Digital integration you wish to emulate.



TDM > General

Avaya Definity

• Leave Turn MWI on FAC and Turn MWI Off FAC blank.

Mitel

- Set MWI values to:
 - TURN MWI On FAC to #21
 - \circ $\;$ TURN MWI Off FAC to #22 $\;$

TDM General Settings					
* PCM Coding	uLaw 👻				
Minimum Call Party Delay (ms)	0				
Maximum Call Party Delay (ms)	2000				
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				

VOIP > General (default)

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 (default)

VoIP Network Group Configuration					
Network Group					
Network Group Label	Network Group #1				
Transport					
Transport Protocol	UDP	-			
SIPS URI Scheme	No	-			
URI Paramete	rs				
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_SRTP +				
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	Off				
* RTP Source UDP Port Validation	Off				
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 (default)

VoIP Authentication					
Inbound VoIP [Server] Out	tbound VoIP [Client]				
	T-L				
	Inbound VolP Configurat	tion			
Inbound Authentication Enabled		No 👻			
Gateway Realm		default.gw.com			
Algorithm		MD5 🗸			
	Methods to Challenge				
Invite Register	Notify 🔲 Info 🔲 Bye 🔲	Refer Options			
	Users				
Realm	User Name	Password			
Add Entry					

PBX Configuration

Avaya

Station Programming

Avaya Digital Set types must be set as 7434ND types

add station COEO1		-	1.0.000	-1	- C	c	
add station 62501		E	age	1	OI	ю	
		STATION					
Extension: 62501		Lock Messages? n		1	BCC:	0	
Type: 7434ND		Security Code:			TN:	1	
Port: 01A0801		Coverage Path 1:		(COR:	1	
Name: DuVoice Digital	#1	Coverage Path 2:		(COS:	1	
		Hunt-to Station:					
STATION OPTIONS							
		Time of Day Lock Table	:				
Loss Group:	2	Personalized Ringing Pattern	1: 1				
Data Module?	n	Message Lamp Ext	: 62	2501			
Display Module?	У						
Display Language:	english	Coverage Module	? n				
Current and La COD.	1	Madda Gamalan But					
Survivable COR:	internal	Media Complex Ext					
Survivable Trunk Dest?	У	IP SoftPhone	? n				
		Remote Office Phone	? N				

On page two of the Digital extension for the DMG you must disable the "LWC reception" then enable the "LWC activation" and enable the "Display Client redirect"

add station 62501	Page 2 of	6
	STATION	
FEATURE OPTIONS		
LWC Reception: none	Auto Select Any Idle Appearance?	n
LWC Activation? y	Coverage Msg Retrieval?	У
LWC Log External Calls? n	Auto Answer:	none
CDR Privacy? n	Data Restriction?	n
Redirect Notification? y	Idle Appearance Preference?	n
Per Button Ring Control? n	Bridged Idle Line Preference?	n
Bridged Call Alerting? n	Restrict Last Appearance?	У
Active Station Ringing: single		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	d	
Multimedia Mode: basic		
MWI Served User Type:	Display Client Redirection?	Y
AUDIX Name:	Select Last Used Appearance?	n
	Coverage After Forwarding?	S
	Multimedia Early Answer?	n

Button Programming

1: call-appr 2: call-appr 9: lwc-store 10: lwc-cancel

Meridian

For each of the ports connected to the DMG configure them as show below (LD 11).

TN	
TYPE	2616
CDEN	8D
CUST	0
FDN	
TGAR	0
LDN	NO
NCOS	7
RNPG	0
SCI	0
SSU	
CLS	FBD, WTA, MTD, FNA, HTA, ADD, HFD, MWA, CNDA, CPFD, CPTD
HUNT	XXXX
LHK	0
KEY	
	00 SCR YYYY (Call Appearance)
	07 PROGRAM
	14 MCK (Message Cancellation)
	15 TRN (Transfer)

DuVoice Configuration

Lines

- Set the extension number for each line to a matching number associated with an extension located on the DMG.
- When connecting to an Avaya PBX be sure to configure **Port number for MWI use** to **Same**.
- Set the hunt group field to the hunt group number.
- Do not check the Register field for any line.

SIP Configuration

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

Set MWI method to SIP.

- Set retries to at least 5 in MWI on/off template.
- Set retry interval to 1 or above. in MWI on/off template.

Testing

DMG Testing

The DMG has the ability to test all the channels including integration, mwi and transfers. These tests are located under Diagnostics.

Test channels

Dianostics > Tests > TDM Self Test

Enter your extensions in the space provided and click Start Test.

TDM Self Verification Test Configuration						
Test Selection			✓ Initiate Call / Answer Call □ Transfer Call			
Test Mode			Sequential Simultaneous			
Call Test Configuration						
Channel Extension Numbers						
	In	iterface				
	Port	Extension				
	1					
	2	[
	з					
	4					
	5					
	6					
	7					
	8					
Clear Auto Fill						

Here you can see sample results from the test. A Green box with the letter P means the test passed. If any box shows red with an F then the test failed.

	TDM Self Verification Test Status															
	Port	Chan				Initia	te Call			Answ	er Call		1	Transf	er Call	
			Status	Outbound Route	Orig	Prog	DTMF	Disc	Ans	CPID	DTMF	Rls	Ans	CPID	DTMF	Rls
	1	1	Complete	1:1 5104->2:1 5105	Р	Р	Р	Р	Р	Р	Р	Р	-	-	-	-
	2	1	Complete	2:1 5105->1:1 5104	Р	Р	Р	Р	Р	Р	Р	Р	-	-	-	-

Test MWI

The DMG can test both setting and clearing an MWI. This will confirm the PBX configuration for lighting MWI's.

Diagnostics > Tests > TDM

- 1. Choose **Send Message**
- 2. Enter the station number in the **Destination Number** field.
- 3. Choose **Set** or **Clear**.
- 4. Click **Start Test**.

TDM Test Configuration					
Test Selection	Initiate Call Send Message				
Port	Automatic 🗸				
Channel	Automatic 🗸				
Destination Number					
Source Name					
Source Number					
Device	TDM				
Message Waiting Status	🖲 Set 🔘 Clear				