



Dialogic[®] 2000 Media Gateway Series Installation and Configuration Note

for Microsoft[®] Lync 2010

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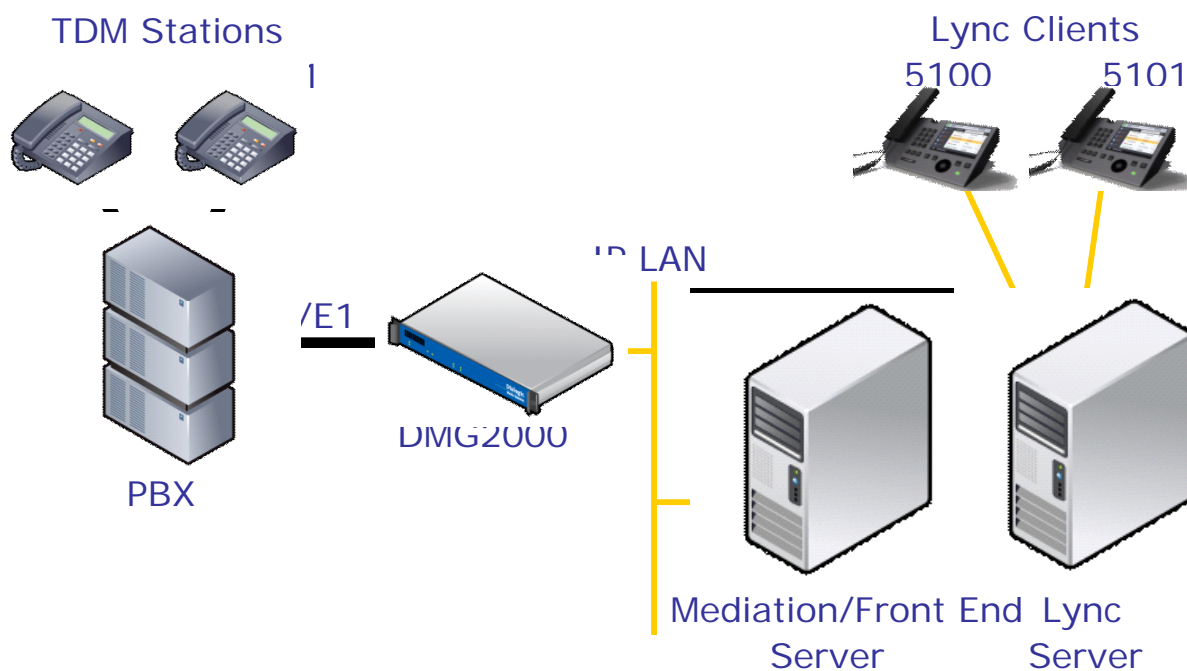
1. Scope

This document is intended to detail a typical installation and configuration of Dialogic® 2000 Media Gateway Series (DMG2000) when used to interface between a PBX or Central Office and Microsoft® Lync 2010.

2. Gateway Configuration Details

Gateway Model(s)	Dialogic® 2000 Media Gateway Series (DMG2030DTIQ, DMG2060DTIQ, DMG2120DTIQ, DMG2060DTISQ, DMG2120DTISQ)
Software Version(s)	Version 6.0 Service 6.0.004_B003
Protocol(s)	T1 CAS, T1/E1 ISDN

2.1 System Diagram



3. Prerequisites

3.1 Trunk Prerequisites

The T1 or E1 trunk must be configured for T1 (NI2, QSIG, 5ESS or DMS protocols) or E1 (Euro ISDN or QSIG protocols).

If connecting to a PBX, refer to the Unified Messaging Integration Notes for guidance at: <http://www.dialogic.com/support/helpweb/mg/iw1904.aspx>

For more information on PBX programming, refer to: <http://www.dialogic.com/solutions/unified-communications/microsoft-pbx-interop-and-config-guides.aspx>

3.1.1 Cabling Requirements

Cabling for ISDN connections must be CAT5e or better. Standard voice quality cable will not provide optimum signal quality, and will cause the gateway to experience problems establishing a connection on the D-Channel.

If connecting the gateway directly to the PBX, it may be necessary to use a T1 crossover cable (Pin 1 is connected to Pin 4 and Pin 2 is connected to Pin 5). This is dependent on the PBX type to which the gateway is being connected, so if after connecting the gateway to the PBX the Alarm light on the front of the gateway remains “Red”, try using the crossover cable.

3.2 Gateway Prerequisites

The DMG2000 must be installed with Version 6.0.SU6.0.004_B003.

4. Summary of Limitations

No limitations noted as of the last update to this document.

5. Gateway Setup Notes

5.1 Initial Setup

There are two options for performing the initial configuration of the DMG2000 – serial or IP. It is recommended that serial be used.

During the initial setup of the DMG2000, you will be setting the following parameters:

- Assign the gateway a unique IP address, subnet mask and IP network gateway address (if the latter is required).

- Configure the gateway to use the SIP VoIP protocol.
- Set the Line Mode to T1 or E1.
- Set the Protocol to match your trunk setting.

Connecting to Console Port

Serial:

Connect to the COM 2 port on the DMG2000 using an appropriate program, such as HyperTerminal.

The serial port configuration is:

Baud rate: 115200
Data Buts 8
Stop Bits: 1
Parity: none
Flow Control: none

Serial Port Pin Out

Pin	Signal
1	Data Carrier Detect
2	Transmit Data
3	Receive Data
4	Data Terminal Ready
5	Signal Ground
6	Data Set Ready
7	Clear to Send
8	Request to Send
9	Ring Indicator

Ethernet:

Default DMG2000 IP address: **10.12.13.74**

Change the IP address of your computer to be in the same subnet as the default. Telnet into the DMG2000 and then follow the commands noted below:

Running QuickCFG

Press Enter key until you get to the "PIMG" prompt. Follow the steps below and modify the settings in **red** to match your environment. The values in **bold** are what you will be entering.

```
PIMG> pwd
Enter Password: IpodAdmin
Admin level accepted.
PIMG-admin> quickcfg
LAN 1 IP Address[10.12.13.74] : (Enter new IP Address that matches gateway IP address entered in Mediation server)
LAN 1 Subnet Mask[255.255.255.0] : (Enter new Subnet Mask)
LAN 1 Default Network Gateway Address[10.2.2.5] : (Enter new Default Network Gateway Address)
LAN 2 IP Address[10.2.2.2] :
LAN 2 Subnet Mask[255.255.255.0] :
Select Line Mode ...
Valid entries:
  1. T1
  2. E1
Enter Number for Line Mode Selection [T1] : 1
Select Protocol ...
Valid entries:
  1. CAS - Loop Start
  2. CAS - Ground Start
  3. CAS - E&M Immediate
  4. CAS - E&M Delay
  5. CAS - E&M Wink
  6. ISDN - QSIG
  7. ISDN - NI-2
  8. ISDN - 5ESS
  9. ISDN - DMS-100
Enter Number for Protocol Selection [ISDN - NI-2] : 6
Saving parameters now...
Parameters successfully configured!
***** Restart Required ***** (Type 'restart')
PIMG-admin> restart
rebooting...
```

5.2 Final Web Configuration

Once the initial configuration is complete and the DMG2000 has been reset, it is necessary to log into the web interface by browsing to the IP address of the DMG2000 using “admin” as the user and using the gateway password (“IpodAdmin” is the default).

Sample Configuration files

Download available at:

http://www.dialogic.com/~media/microsoftuc/DMG2000_Lync_Config.zip

- There are two sample configuration files for adjusting gateway parameters to enable interoperability with Microsoft® Lync 2010 specific settings. The “**DMG2000 Lync Config u-law.ini**” file, among other functions, will set up the codec to be G.711 uLaw for use in North America and other areas that use uLaw. The “**DMG2000 Lync Config a-law.ini**” file, among other functions, will set up the codec to be G.711 aLaw for use in Europe and other regions that use aLaw. In addition to changing the default settings, these files also contain some sample Dial Plan configurations to provide a starting point for configuration the E.164 phone number conversions.
- To import one of these files into the DMG2000, use the Import/Export page on the DMG2000 web interface. Select **IMPORT** and browse to reach the desired “.ini” file to import.

Setting T1 / E1 Parameters

Line Encoding and Framing

It is necessary to ensure that the Line Encoding and Framing match those of the T1 or E1 line coming from the PBX or Central Office.

ISDN Protocol

Since ISDN protocols are not necessarily symmetric protocol it is necessary to determine which device (PBX or Gateway) is going to function as the Network side of the protocol. It is recommended that if the PBX can support Network side that it be configured as Network in which case the gateway needs to be configured as Terminal. Setting the Protocol side is done by changing the “Telephony Port Interface Side” to the setting you want the gateway to function.

Setting Line Encoding, Framing, and ISDN Protocol

Navigate to the **TDM > T1/E1** page.

The screenshot shows the Dialogic configuration interface for T1/E1 settings. The interface is divided into a sidebar on the left and a main configuration area on the right. The sidebar contains navigation options such as Status, Configuration, Diagnostics, and System. The main configuration area is titled "T1/E1 Configuration" and contains several sections:

- T1/E1 Port Selection:** Select Port to Modify: all ports
- Line Settings:**
 - * Line Mode: T1
 - * Signaling Mode: ISDN
 - * Telephony Port Interface Side: Terminal
- T1 Line:**
 - * Line Encoding: B8ZS
 - * Framing: ESF
 - * Selects Transmit Pulse Waveform: Short_Haul_110ft
- T1 ISDN protocol:**
 - * ISDN Protocol: NI-2
 - ISDN Protocol Variant: None
- General ISDN Settings:**
 - QSIG Protocol Specification: ISO
 - Network-Specific Facilities (NSF): None
 - ISDN Answer Supervision Enable: Yes
- Failover Settings:**
 - * Enable Failover: No

Two callout boxes with arrows point to the "Telephony Port Interface Side" and "Line Encoding" dropdown menus. The first callout box contains the text "Set ISDN Network or Terminal side." and the second callout box contains the text "Set T1/E1 Line Encoding and Framing." Below the configuration area are "Submit" and "Cancel" buttons.

Setting PCM Coding for TDM

Navigate to the **TDM > General** page.

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 ▼

aLaw – E1
uLaw – T1

Setting VoIP Parameters

Navigate to the **VoIP > General** page and the following VoIP parameters will need to be set:

- Gateway's FQDN
- DNS server IP addresses
- TCP or TLS Listen ports
- DNS monitoring if DNS load balancing is being used

Voip General Settings	
User-Agent	
* Host and Domain Name	gateway.test.local
Call as Domain Name?	No
Invite Expiration (sec)	120
Reliable Provisional Responses	Supported
Server	
* DNS Server Address	192.168.1.5
* DNS Server Address (Secondary)	192.168.1.6
DNS Translation of Phone Numbers	No
TCP/UDP	
* UDP/TCP Transports Enabled	No
* TCP/UDP Server Port	5068
TCP Inactivity Timer (sec)	120
TLS	
* TLS Transport Enabled	Yes
* TLS Server Port	5067
* SSL/TLS Protocol	SSLv3_TLSv1
* Mutual TLS Authentication Required	Yes
TLS Inactivity Timer (sec)	120
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	Yes
QoS	
* Call Control QoS Byte	0

Enter the FQDN assigned to the gateway.

Ensure that Reliable Provisional Response is set to Supported.

Enter the IP address of a DNS server and a backup if available.

Enable TCP and or TLS for the gateway to listen on. Configure the Port to match that defined in Topology Builder.

Make sure that the TCP and TLS Inactivity Timers are set to 120.

If using DNS load Balancing to distribute the calls between multiple Frontend Servers set this to Yes.

Setting Network Groups

Navigate to the **VoIP > Network Groups** page.

On the Network Groups page you will be configuring the outbound transport, voice codec, packet size and whether you want RTP or SRTP. It is possible to set up multiple different network groups with different characteristics for different IP endpoints. The Networks groups will be referenced in the route table setup further down in this document.

VoIP Network Group Selection	
Select Group to Modify	Network Group#1

VoIP Network Group Configuration	
Network Group	
Network Group Label	Network Group # 1
Transport	
Transport Protocol	TLS
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
Codec #4	None
Packet Time (ms)	20
SRTP	
SRTP Preference	SRTP_Only
Authentication Tag Length	80
MKI on Transmit Stream	No
Key Derivation Enable	No
Key Derivation Rate	16
Window Size Hint	64
UnEncrypted SRTP Enable	No
UnEncrypted SRTCP Enable	No
UnAuthenticated SRTP Enable	No

Select the transport for outbound calls from the gateway.

If using SRTP then set SRTP Preference to SRTP_Only.

Set MKI on Transmit Stream to No.

Configuring Media Settings

Navigate to the **VoIP > Media** page.

On this page, you will be setting up Early Media, configuring fax for G.711 pass-through, setting the DTMF digit transmission to RFC2833 with the lowest latency possible. You will also be configuring the RTP port validation.

VoIP Media Settings	
Early Media	
RFC 3960 Early Media Support	Always
Send Early 183 Progress Response	Yes
Require Reliable Provisional Responses	Yes
Audio	
Signaling Digit Relay Mode	Off
Voice Activity Detection	On
Acceptable Media	RTP_S RTP
Continue Ringback on CN	Yes
Packet Time (ms) for Inbound VoIP	20
Digit Relay Mode	RFC2833-LowLatency
Telephone-Event Payload Type	101
* G722 Enable	No
Fax	
Fax IP-Transport Mode	G.711-Passthrough
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

Enable Early Media Sending the 183 messages and Reliable Provisional Responses.

Set the Digit Relay mode to RFC2833-LowLatency to minimize latency.

Set the Fax mode to G.711-Passthrough.

Turn off both RTP port and address validation.

Configuring DSP Settings

Navigate to the **DSP Settings** page.

DSP Advanced Settings	
Line Echo Cancellation	On
Line Echo Cancellation NLP (non-linear processor)	On
Line Echo Cancellation NLP Aggressiveness	Normal
Echo Cancellation Gain (dB)	0
Voice activity noise floor (dBm)	-32
Voice activity Measurement Period (ms)	200
Voice activity Signal to Noise Ratio (dB)	20
Call Progress Filter Threshold (dBm)	-30
Call Progress Filter Debounce (ms)	100
Call Progress Filter SNR (dB)	20
Call Progress Filter Two Tone Max Twist (dB)	3
Jitter-Buffer Minimum Delay (ms)	0
Jitter-Buffer Maximum Delay (ms)	200
Jitter-Buffer Initial Delay (ms)	20
Jitter-Buffer Adaptation Period (ms)	10000
Jitter-Buffer Deletion Threshold (ms)	500
* Jitter-Buffer Frame Deletion Mode	Soft
TDM to TDM Media Always Clear Mode	Off
TDM to IP Gain Adjustment (db)	0
IP to TDM Gain Adjustment (db)	0

Enable Echo Cancellation with NLP and normal Aggressiveness.

Configure the Jitter Buffer.

AGC	
* AGC Maximum Noise Floor (dBm)	-21
* AGC Minimum Allowable Signal (dBm)	-45
* AGC Minimum SNR Requirement (dB)	12
* VAD On-to-Off Hangover Interval (ms)	500
AGC G.169 Compliance	Off
IP to TDM AGC Enable	Off
IP to TDM AGC Target level (dBm)	-14
IP to TDM AGC Max Gain (dB)	12
IP to TDM AGC Min Gain (dB)	-15
TDM to IP AGC Enable	Off
TDM to IP AGC Target level (dBm)	-14
TDM to IP AGC Max Gain (dB)	12
TDM to IP AGC Min Gain (dB)	-15

Turn AGC off in both Directions.

Configuring Route Table for E.164 support

Configuring Calls to PSTN Network

Microsoft® Lync sends the phone number to the DMG2000 in E.164 format. It is necessary to convert this to a dialable number, which, in turn, can be sent to the central office or to your PBX.

North America (DMG2000 Lync Config u-law.ini)

- The rules in the sample configuration file “DMG2000 Lync Config u-law.ini” perform the following dial sting manipulations:

Phone number type	Microsoft® Lync sent phone number	Phone number sent to CO or PBX
4-digit extension from non-DID number	+14258811000;ext=1234	1234
4-digit Extension (starting with 881)	+14258812345	2345
Local call	+14251234567	91234567
Long Distance	+14151234567	914251234567
International	+44123456789	901144123456789

- It will be necessary to modify these rules to fit your environment. You will need to match your internal extension pattern.
- If you are connected directly to a trunk line coming from a phone company, it will be necessary to remove the trunk access code “9” from these rules. Depending on your PBX configuration, it might be necessary as well to change the sample trunk access code to one that matches your environment.
- To change the Routing Table settings, log into the web interface, then open the “Routing Table”. With the Radio button selected for Inbound VoIP rules select among the five “Inbound VoIP Rules” to see the routing and CPID manipulations for each rule.
 - In the “non-DID internal” rule the gateway will automatically extract the number string following the “ext” field of the Invite and pass to the Routing table. Because of this all that is necessary to do in the CPID manipulation is to pass the extension.
 - The “Internal” rule will extract the last 4 number from the right side to create a 4-digit extension to be dialed on the local PBX.

- The “Local” rule is extracting seven (7) digits from the right side (relx(d,7) and pre-pending the trunk access code “9” to the number. If you are connected directly to the phone company, you will want to change this rule just to extract the seven (7) digits. It should look like this: rext(D,7).
- All the “Domestic” rule is doing is adding the trunk access number. To the 11 digits following the “+”. Depending on your configuration, this can be changed or deleted.
- The “International” rule is adding the trunk access number “9” and the international access number “011”. If necessary, you can change or delete the trunk access number.

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Config > Routing Table Ports

Router Configuration

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

Select	Enable	Rule Label	Request Type	Originating VoIP Host Address
<input type="checkbox"/>	<input checked="" type="checkbox"/>	non-DID internal	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Internal	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Local	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Long Distance	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	International	Any	*

Detailed Configuration for Inbound VoIP Rule: **non-DID internal**

Inbound VoIP Request Matching			
CPID Matching			
Calling Number	*	Called Number	xxxx
Calling Name	*	Called Name	*
Redirect Number	*	Redirect Name	*

Outbound Routes			
Device Selection			
Outbound Destination	TDM	Trunk Group	Outbound trunk
CPID Manipulation			
Calling Number	S	Called Number	rext(D,4)
Calling Name	S	Called Name	D
Redirect Number	R	Redirect Name	R

Select Primary / Alternate Route

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

Europe (DMG2000 Lync Config a-law.ini)

Phone number type	Microsoft® Lync sent phone number	Phone number sent to CO or PBX
National	+44123456789	0123456789
International	+49123456789	0049123456789

- Changing “National” dial plan to match your country
 - In this example, and in the applicable attached sample “.ini” configuration file, the dial plan is set up to determine a “National” number in the UK. If you are located in another country, you will need to change the National dial plan to match your specific country code. To do so, you will need to edit the “Inbound VoIP Routing” rule and modify the “CPID Manipulation” setting to match your country.
- Inbound VOIP Routing
 - The Inbound routing rules compare the number sent by Microsoft® Lync and look for the country code. If the located code matches the desired code for the country in which the DMG2000 is deployed - in this case the UK - then the manipulation for a national number is run. All other calls are treated as international calls.
- CPID Manipulation
 - For a “National” number, the country code is replaced with a “0” and the number is dialed. For international numbers, a “00” is added to the beginning of the dial string.
 - If the DMG2000 is deployed behind a PBX, it might be necessary to add a trunk access to the dial strings as well. To accomplish this, the “Called Number” field of the CPID Manipulation section will need to add the trunk access code. In the example below, you would need to change the rule to be:
 - National repl(D,“+44”,“90”)
 - International repl(D,“+”,“900”)

Router Configuration

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

Inbound VoIP Rules

Select	Enable	Rule Label	Request Type	Originating VoIP Host Address
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Local	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	International	Any	*

Add Rule Delete Rule

Detailed Configuration for Inbound VoIP Rule: Local

Inbound VoIP Request Matching

Calling Number	*	Called Number	+44	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes

Device Selection

Outbound Destination	TDM	Trunk Group	Any Trunk
----------------------	-----	-------------	-----------

Calling Number	S	Called Number	repl(D,"+44","90")	Redirect Number	R
Calling Name	S	Called Name	*	Redirect Name	R

Select Primary / Alternate Route

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

Delete Delete Delete Delete Add Alternate Route

Submit Cancel

Set the Local Country code.

Remove the country code and replace with "09".

Configuring Calls from TDM to Microsoft® Lync Server

Directing PSTN / PBX Calls to Mediation Server

- It is necessary to direct all incoming calls from the TDM (PSTN) side to the mediation server. This is done in the Routing Table tab of the web interface. Select the Inbound TDM Call Routing tab and enter the IP-address or FQDN of the Mediation server or pool of servers in the URI field.
- Hit "Submit" to save the changes.

The screenshot shows the 'Router Configuration' web interface. At the top, there are tabs for 'Inbound TDM Rules', 'Inbound VoIP Rules', 'TDM Trunk Groups', and 'VoIP Host Groups'. The 'VoIP Host Groups' tab is active, displaying a table with the following data:

Select	Name	Load-Balanced	Fault-Tolerant	Network Group	
<input type="checkbox"/>	VoipGroup-1	false	false	Network Group #1	1

Below the table are 'Add Host Group' and 'Delete Host Group' buttons. A callout box points to the 'VoipGroup-1' name field with the text: 'Name of VoIP Endpoint (needed for setting up route)'. Another callout box points to the 'Network Group #1' dropdown with the text: 'Select the Network Group created in the previous steps.' Below the table, a section titled 'The selected Host Group is referenced by the following rules:' shows a list of rules, including '[inbound TDM] InboundTdm (Primary Route)'. To the right, a 'Host List' table shows the configuration for the selected group:

Host	
pool.test.local:5067	Delete

A callout box points to the 'pool.test.local:5067' entry with the text: 'Set IP address of Mediation server or the FQDN for the pool of mediation servers. Be sure to include the port that you want to transmit to. In this example 5067.' There is also an 'Add Host' button below the Host List.

Setting Up Route from TDM to IP

- Now you will need to select the "Inbound TDM Rules" to create a route for the incoming TDM calls to the VoIP endpoint:
- Select the "Inbound TDM Rules" radio button and then select the first "Inbound TDM Rule".
- In the "Outbound Routes" for this rule select the "Outbound Destination" and set it to "VoIP".
- For the "Host Group" select the name that was created while setting the IP address in the last step.
- Hit "Submit" to activate the rule.

The screenshot shows the Dialogic Router Configuration web interface in a Windows Internet Explorer browser window. The browser address bar shows `http://192.168.1.200/wm/Default.htm`. The page title is "Config > Routing Table".

The main configuration area is titled "Router Configuration" and includes radio buttons for "Inbound TDM Rules", "Inbound VoIP Rules", "TDM Trunk Groups", and "VoIP Host Groups". The "Inbound TDM Rules" section contains a table with the following data:

Select	Enable	Rule Label	Request Type	Trunk Group
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Internal call	Any	Any Trunk
<input type="checkbox"/>	<input checked="" type="checkbox"/>	External call	Any	Any Trunk

Below the table are "Add Rule" and "Delete Rule" buttons. The "Detailed Configuration for Inbound TDM Rule: Internal call" section includes "Inbound TDM Request Matching" and "Outbound Routes" sections.

The "Outbound Routes" section includes "Device Selection" and "CPID Manipulation" sections. The "Device Selection" section has "Outbound Destination" set to "VoIP" and "Host Group" set to "VoipGroup-1". The "CPID Manipulation" section has "Calling Number" set to "S", "Called Number" set to "D", "Redirect Number" set to "R", "Calling Name" set to "", "Called Name" set to "D", and "Redirect Name" set to "R".

At the bottom of the "Outbound Routes" section, there are radio buttons for "Primary", "Alt-1", "Alt-2", "Alt-3", and "Alt-4", with "Primary" selected. There are also "Delete" buttons for each alternate route and an "Add Alternate Route" button.

Two callouts with arrows point to the "VoIP" and "VoipGroup-1" fields. The first callout says "Set Outbound Destination to be VoIP." The second callout says "Name of VoIP group containing Mediation server address." "Submit" and "Cancel" buttons are located at the bottom of the configuration area.

Converting the Called Party to E.164 format

- It is not necessary to convert the Called Party number to E.164 before it is sent to Microsoft® Lync as the existing normalization rules in Microsoft® Lync will take the number the gateway is receiving from the PBX normalize it to the correct format.

Converting Calling Party number to E.164 format

- If you want Microsoft® Lync to be able to do a reverse look up in the GAL or a users contact list it is necessary to have the gateway convert the Calling Party number to E.164 to match the format in the GAL. This manipulation of the calling party information will be dependent on the information that the gateway is receiving from the PSTN or PBX. Here are a few examples of necessary conversions and the rule to achieve the needed conversion.

PSTN / PBX Sends	Manipulation Needed	Dial Plan Rule
4-digit extension Ex. 1234	Need to add +1 (area code) and first three number of phone number +1425882	" +1425882" +S
7-digit phone number Ex. 8821234	Need to add +1 and the area code. +1425	" +1425" +S
10-digit number Ex. 428821234	Need to add the +1	" +1" +S
Local Call in UK Ex. 1628123456	Need to add +44	" +44" +S

- If you are unsure what your PBX or the PSTN will be sending to the gateway, you can place a couple of test calls into the gateway and look at the call log to see the information that is being sent. The calling party information can be seen in this screen shot which shows a call from extension 1111 to 425-882-1234. Be sure to place calls from both on the PBX and from outside. It is possible that you will receive a 4-digit number from inside and a 10-digit number from outside. If this is the case you will need to enter two rules one for each number format.

Call Log						
ID	Start Time	End Time	Source	End Reason	Inbound info	Outbound info
3	5/7 17:00:06	5/7 17:00:14	From TDM Network	TDM: Normal	1:1111,->4258821234,->, [Rsn=Direct]	+14258821234,,,10.10.10.1
2	5/7 16:59:37	5/7 16:59:44	From TDM Network	TDM: Normal	1:1111,->4258821234,->, [Rsn=Direct]	+14258821234,,,10.10.10.1
1	5/7 16:58:57	5/7 16:59:07	From TDM Network	TDM: Normal	1:1111,->4258821234,->, [Rsn=Direct]	+14258821234,,,10.10.10.1

Calling Party number

Called Party number

- Once you have determined the format of the calling number for your environment and matched it to the rule in the above table, you will need to enter the rule into the DMG2000 gateway. This is done by selecting the Inbound TDM Rules radio button on the Routing Table web page and modifying the "Outbound Routes" rule for "Calling Number" from the samples provided in the configuration files.
 - In the "Calling Number" box enter the rule from the table above that has been changed to match your environment.
 - Hit "Submit" to save the rule.

The screenshot shows the Dialogic web interface for configuring a Router. The main area is titled "Router Configuration" and includes sections for "Inbound TDM Rules", "Inbound TDM Request Matching", and "Outbound Routes".

Inbound TDM Rules Table:

Select	Enable	Rule Label	Request Type	Trunk Group
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Internal call	Any	Any Trunk
<input type="checkbox"/>	<input checked="" type="checkbox"/>	External call	Any	Any Trunk

Outbound Routes - CPID Manipulation Table:

Calling Number	Called Number	Redirect Number
" +1425882" + S	D	R
S	D	R

A callout box with an arrow points to the "Calling Number" field in the Outbound Routes section, containing the text: "Enter Calling Party Manipulation."

6. Troubleshooting

6.1 Debugging Tools

- `Ethereal/Wireshark` – Used to view and analyze the network captures provided by the DMG2000 diagnostic firmware.
- `Adobe Audition` – Used to review and analyze the audio extracted from the network captures in order to troubleshoot any audio-related issues.

7. Gateway Trace Masks

These keys are helpful during troubleshooting scenarios and should be considered as keys to activate by default for all troubleshooting.

- `voip prot` and `voip code` – These allow the collection of all SIP-related messages as they are sent from and received by the DMG2000. Such data can be important, as in cases where you believe the DMG2000 is unable to communicate properly with the messaging server.
- `tel event` and `tel code` – These allow the collection of all circuit-side activity. Such data can be important, such as when addressing the following scenarios:
 - Call control problems (dropped calls, failing transfers, etc.)
 - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc.)
- `teldrv prot` – This allows the collection of all ISDN messages both transmitted and received on the DMG2000 front end interface. This data can be very important, such as when addressing the following scenarios:
 - Call control problems (dropped calls, failing transfers, etc.)
 - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc.)
- `RouteTable (all keys)` – This will allow the collection of events around the parsing of the dial plan. This will enable the user to see the calling and called numbers being sent to the gateway and see how they being changed by the dial plan configuration.

The following keys are helpful when confronting certain issues and can be enabled for targeted troubleshooting of specific problems. Activating these keys may generate large amounts of data on busy systems and increase the size of the collected log files, but doing so will not harm system performance.

- `dspi f` (all keys) – This allows the collection of tone related data. Such data is very helpful in cases where you think you have problems relating to detection specific tones that should be, should not be, or are expected to be present at specific times during the call. If you do not suspect a tone related issue, this key may be left disabled.

NOTE: Turning on all traces is not recommended. Doing so floods the debug stream with significant amounts of information that can cause delays in determining the root cause of a problem.

Additional information regarding the Dialogic® 2000 Media Gateway Series (DMG2000) can be found at the following link: <http://www.dialogic.com/support/helpweb/mg/default.aspx>