

# Dialogic<sup>®</sup> 4000 Media Gateway Series Integration Note

NEC NEAX2400 IPX

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## 1. Scope

This document is intended to detail a typical installation and configuration of the Dialogic<sup>®</sup> 4000 Media Gateway Series if used to interface between a PBX and the Microsoft<sup>®</sup> Office Communications Server (OCS) 2007 application.

## 2. Configuration Details

Listed below are details of the PBX and gateways used in the testing on which this document is based.

## 2.1 PBX

PBX Vendor	NEC
Model(s)	NEAX2400 IPX
Software Version(s)	Ver.17 Rel.03.46.001
Additional Notes	N/A

## 2.2 Gateway

Gateway Model	Dialogic <sup>®</sup> 4000 Media Gateway Series
Software Version(s)	Dialogic <sup>®</sup> Diva <sup>®</sup> System Release software version 8.3.2 build 459 (formerly called Diva <sup>®</sup> Server software) Dialogic <sup>®</sup> Diva <sup>®</sup> SIPcontrol <sup>™</sup> Software version 1.6 build 46 (DSSIPControl.msi)
Protocol	T1 Q.SIG

## 2.3 System Diagram

The diagram below details the setup used in the testing and creation of this document. In the diagram, the abbreviation DMG4000 stands for the Dialogic<sup>®</sup> 4000 Media Gateway Series and OCS Server stands for Microsoft<sup>®</sup> Office Communications Server (OCS) 2007.



## 3. **Prerequisites**

## 3.1 **PBX Prerequisites**

The PBX must have all supplemental service packages installed for the Q.SIG protocol to operate properly and to provide all advanced supplemental services.

## 3.1.1 PBX Equipment Required

To support the T1 Q.SIG configuration as documented you need a PA24PRTB-A ISDN T1 line card.

## 3.1.2 PBX Cabling Requirements

The cabling for Q.SIG connections must be CAT5e or better. A standard voice quality cable will not provide the desired signal quality and will cause the gateway to have issues establishing a connection on the D-channel.

## 3.2 Gateway Prerequisites

The gateway needs to support a T1 Q.SIG interface.

## 4. Summary of Limitations

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

4

#### **Gateway Setup Notes** 5.

Steps for setting up the gateway:

- Configuration of the Dialogic<sup>®</sup> Diva<sup>®</sup> Media Board drivers.
   Configuration of the Dialogic<sup>®</sup> Diva<sup>®</sup> SIPcontrol<sup>™</sup> software.

#### Dialogic<sup>®</sup> Diva<sup>®</sup> Media Board Configuration 5.1

The Diva Media Boards are configured in the Dialogic<sup>®</sup> Diva<sup>®</sup> Configuration Manager. To open the Configuration Manager, click:

Start > Programs > Dialogic Diva > Configuration Manager.

Note: In the Dialogic<sup>®</sup> Diva<sup>®</sup> software and documentation, Diva Media Boards are referred to as Diva Server adapters.

A screen similar to the one below will appear.

👰 Active Configuration - Diva Server Configuration Manager			×
Elle Edit Insert View Tools Help			
🗅 🚅 🔜 🔥 🖌 Ioobar 🛛 😵 😥			
✓ <u>Status</u> Bar	Property	Value	
Advanced	Line Type	Primary Rate Line (23 B-Channels)	
	Switch Type	Q-SIG T1 )	J
Services diva	PBX Type	Generic	
	Q-Sig Standard	Automatic	
9999	Call Reference Format	Standard	
	Interface Type	Point-to-Point (Standard)	1
	Direct Dial In (NT2)	No	1
V-4PRI	Number Type	Range of Extensions	
Adapters	Lowest Extension	000	1
() mura	Highest Extension	999	
ŶŶŶŶ	TEI	0	
	Layer 2 Connect Mode	Permanent	
1 2 3 4	Voice Coding	Protocol Default	
	Operation Mode	TE - Terminal Equipment (Recommended)	
Lines ( 🔊 🛇	Fractional Line	No	
	Generate Ring Tones	Yes	
	Device Mode	Standard	1
	Encoded Signal Power Limiter	Protocol Default	1
	Disconnect on Progress	Protocol Default	1
	Transparent Facility	Off	
	Rerouting	On	
	PR Invite	Off	
	Redirecting Number Emula	Disabled	
	DTMF Clamping	Off	
	Recording AGC	Off	1
	Dial Pulse Detection	Off	
	ECT Emulation	Disabled (Handled by Network)	
	Limit Call Rate	Off	
	See legal notice in Reference	Guide.	
	Configure the line-specific pr	operties here.	
	To assign the configured dire services and adapters.	ectory numbers to the services, select the bindings between	
	Por more and adaptor at		-
Shows or hides advanced settings.			1

Note: The number of TDM circuits varies depending on the used Dialogic<sup>®</sup> Media Gateway model.

For this setup:

- Set the property Switch Type to Q-SIG T1.
- If your PBX does not provide ring tones to callers from TDM, set the property Generate Ring Tones to Yes.

To activate the change, click File > Activate.

Make these configuration changes for each TDM circuit you are going to use on the Dialogic<sup>®</sup> Media Gateway.

## 5.2 Dialogic<sup>®</sup> Diva<sup>®</sup> SIPcontrol<sup>™</sup> Software Gateway Application

The Diva SIPcontrol software is configured via the web based interface. To open the web interface, click Start > Dialogic Diva > SIPcontrol Configuration.

On the main page, click the SIPControl link to display the different configuration menus. The PSTN Interface Configuration section should automatically include all ports detected in the system.

Note: If you do not see any detected ports, you may need to add <u>http://127.0.0.1</u> as a trusted site. From *Microsoft*<sup>®</sup> Internet Explorer, click Tools > Internet Options > Security > Trusted Sites. Use <u>http://127.0.0.1:10005</u> to get to the configuration.

In order for the Diva SIPcontrol software to route calls, the proper routes must be created and configured. Each route consists of a source interface and a destination interface. PSTN controllers and SIP peers are considered either a source interface or a destination interface depending on the call direction.

## 5.2.1 PSTN Interface and Network Interface Configuration

The following is a typical configuration.

Dialogic	Diva@ Configuration - Mic	crosoft Internet Explorer							
<u>E</u> le <u>E</u> di	t <u>V</u> iew F <u>a</u> vorites <u>T</u> ools	Help							
🕒 Back	• 🗇 - 🗷 🗈 🏠 🔎 s	iearch 👷 Favorites 🕢 🍰 🌜							
Address 🧃	http://127.0.0.1:10005/sipco	ntrol.cgi							
Links 🍓 P	BX IP Media Gateway								
Dia	logic								
		HOME	s	Pcontrol Configuration					
		DOTN Interface Configuratio	~						۲
œ	Configuration	Point Interface Configuratio	1						¥
	SIPcontrol >	Name	Nr	Hardware Description	Channels	Dialplan		Enabled	
	Password	Controller1	1	Eicon Diva Server V-4PRI/E1/T1 - PORT 1 SN: 1	223 23	none	*	V	Details
		Controller2	2	Eicon Diva Server V-4PRI/E1/T1 - PORT 2 SN: 1	223 23	none	×	1	Details
	System	Controller3	3	Eicon Diva Server V-4PRI/E1/T1 - PORT 3 SN: 1	223 23	none	•	•	Details
	Service Status	Controller4	4	Eicon Diva Server V-4PRI/E1/T1 - PORT 4 SN: 1	223 23	none	•	<b>v</b>	Details
<b>9</b>	Licensing	Network Interface Configura	tion						۲
	License Management	Name	Devic	e	IP Address	Protocol	SIP L	isten Port	Enabled
		Intel(R) PR01000 EB Network Co	nn Intel(R	) PRD/1000 EB Network Connection with I/O Acceleration	192.168.0.106	al 💌	9803	Σ	₹
		Local Loopback Interface	Local	Loopback Interface	127.0.0.1	al 💌	5060	_	П
		RTP Start Port	30000						
		RTP End Port	39999	3					
		Jitterbuffer Size Min (ms):	0						
		SIP Peer Configuration							۲
		Routing Configuration							•

The Network Interface Configuration will be used by the Diva SIPcontrol software for listening to the SIP traffic from Microsoft<sup>®</sup> Mediation Server. Given that on these gateways the Microsoft<sup>®</sup> Mediation Server component and the Diva SIPcontrol software are running in the same system, you will need to change SIP Listen port to 9803 or to an available un-used port. Later during the Microsoft<sup>®</sup> Mediation Server configuration, you will need to set the PSTN Gateway next hop setting to 9803 to match.

## 5.2.2 SIP Peer Configuration

Create one SIP peer to talk to Microsoft<sup>®</sup> Mediation Server as shown below.

dit SIP Peer Configuration	
Name:	Mediation Server
Peer type:	MS 0 CS 2007 / Mediation Server
Default SIP to PSTN peer:	
Host:	192.168.0.106
Port:	5060
IP protocol:	TCP •
Domain:	
Display name to:	
Display name from:	
User name to:	
User name from:	
Force T.38 reinvite:	Г
Alive check:	Г
Address map inbound:	none
Address map outbound:	none

## 5.2.3 Routing Configuration

In the Routing Configuration section, you must create two routes, one for the inbound direction (TDM to IP) and one for the outbound direction (IP to TDM). Once you have created the routes, click the Save button for the changes to take effect.

IP Peer Configuration											۲
Name	Default SIP to PSTN Peer	Host	Port	IP Protocol	Display Name to	Dialplan		Enabled			
Mediation Server	<b>v</b>	192.168.0.1	06 5060	TCP		none		5	Details	Delete	
			Add								
outing Configuration											۲
Name	Sources		Destinations				Addres	s Map	Enabled		
Outbound Route	Mediation Server	1	Controller1 (Si Controller4 (Si	ave), Controller ave)	2 (Slave), Controller3	(Slave),	none	×	9	t t	Details Delete
Inbound Route	Controller1, Controller2, Cor Controller4	ntroller3,	Mediation Sen	ver (Slave)			none		4	1 1	Details Delete
				a a a l							

## 5.2.4 Number Normalization

The Dialplan Configuration and Address Map Configuration sections are used for manipulating dial numbers. For most PBX dialplans, an address map is required. See the following examples.

## 5.2.4.1 Dialplan Configuration Example

To create a dialplan, click Add from the Dialplan Configuration. The following screens show how to set up a dialplan for a Microsoft<sup>®</sup> Office Communications Server (OCS) 2007 application with the following dialplan from the PBX. (This may not match to the PBX programming in section 6 and the Setup in section 2.3).

Area code: 716 Base number: 639 Extensions: 4 digits Access code: 9

dit Dialplan Configuration	۲
Name:	Dialplan 716
Country code:	1
North-American numbering plan:	
Area code:	716 With national prefix 💌
Other local areas:	
Base number:	639
Maximum extension digits:	4 -
International prefix:	011
National prefix:	1
Access code:	9
PSTN access code provided by SIP caller:	
Incoming PSTN access code provided by PBX:	

Complete the settings and click OK.

For the dialplan to be applied to outbound calls, click the Details button of the PSTN controller and configure the Address Normalization settings as shown in the screen below. This converts the dialed numbers into the format based on the dialplan for the PBX. If the dialed number is for an internal user, it is converted into a 4-digit extension. If the called number is for a national call, 91 is prepended. Click OK on this page, and Save on the next page for the changes to take effect.

http://127.0.0.1:10005 - Edit PSTN Interface	e - Microsoft Internet Explorer
General Configuration	
Hardware description:	Eicon Diva Server V-4PRI/E1/T1 - PORT 1 SN: 1223
PSTN interface number:	1
Name:	Controller1
Address map inbound:	none
Address map outbound:	none
Dialplan:	Dialplan 716 -
Number format (outbound):	Extension
Encoding (outbound):	Use prefixes
Default numbering plan:	unknown
PSTN Call Transfer Settings	
The Call Transfer settings depend on the	e capabilities of the communication platform (PBX, switch).
Blind Call Transfer (A- and C-Party on P	≥STN side)
Transfer type:	With consultation call (Explicit Call Transfer)
Done	

For the dialplan to be applied to inbound calls, click the Details button of the configured SIP peer and configure the Address Normalization settings as in the screen below. This converts the phone number into the E.164 format as needed by Microsoft<sup>®</sup> Office Communications Server 2007. Click OK on this page, and Save on the next page for the changes to take effect.

Address Normalization		
Dialplan:		Dialplan 716 💌
Number format (outbound):		International number
Encoding (outbound):		Use type flag 💌
	OK	Cancel

## 5.2.4.2 Address Map Configuration Example

If the dialplan does not meet your setups special requirements, the Address Map Configuration can be used. An address map entry uses regular expressions (RegEx) (so does Microsoft<sup>®</sup> Office Communications Server 2007) for converting the call address format for inbound or/and outbound direction.

Important note before applying regular expression rules in address maps: The call address for outbound calls (IP to TDM) includes a "@hostname" part. For example, <u>+17166391234@DMG4000.bufocs.local</u> is the call address, not just +17166391234. For inbound calls (TDM to IP), the call address is the called or calling number, with a possible prefix "+", "N", or "S". For example, an inbound call has called number 1234 with ISDN type of numbering flag set to *Subscriber*, and the calling number 49715233334444 with ISDN type of numbering flag set to *International*. The called address will be S1234 and the calling address will be +4971523334444.

If the ISDN type of numbering flag is set to National, the prefix "N" will be used with the call number. If the type is Unknown, no prefix is used.

#### Outbound call example using address maps:

Microsoft<sup>®</sup> Office Communications Server 2007 sends the E.164 dial number format to the SIP gateway. Both called and calling numbers need to be converted into a format that the PBX can accept. If the same PBX dialplan as in the previous section is used, the following conversions are needed.

Calling number	From Microsoft <sup>®</sup> OCS	То РВХ
Internal	+1716639xxxx	716639xxxx

Called number	From Microsoft <sup>®</sup> OCS	То РВХ
To Internal	+1716639xxxx	XXXX
To National	+1xxxxxxxxx	91xxxxxxxxx
To International	+XXXXXX	+XXXXXX

Below is RegEx for the conversion tables above.

Sub rule name	Expression	Format	Stop on match
Calling number	^\+1(716639\d{4})	\$1	Not checked
Called - Internal	^\+1716639(\d{4})	\$1	Checked
Called - National	^\+1	91	Checked
Called - International	^\+	9011	Checked

Below are the configured address maps for outbound calls. The order of the below four sub rules and the stop on match check mark are relevant:

Address Map Configuration				0
Name	Rule Name	Stop on Match	Enabled	
	Calling number		V	↑ ↓ Details Delete
	Called - Internal	<b>v</b>	V	↑ ↓ Details Delete
Outbound	Called - National	<b>v</b>	<b>v</b>	↑ ↓ Details Delete
	Called - International	V	<b>v</b>	↑ ↓ Details Delete
	Add Rule			
		Add		

The following screen shows the first sub rule that converts the E.164 calling number into a 10-digit national number:

Address map name:	Outbound	
Rule name:	Calling number	
Called address expression:		
Called address format:		
Calling address expression:	^+1(716639\d{4})	
Calling address format:	\$1	
Redirect address expression:		
Redirect address format:		
Stop on match:		
NOTE for call address formats: - Addresses received from PSTN (o where <x> represents the number t "S" (subscriber) or empty (unknown - Addresses received from SIP are v headers.</x>	r those normalized via dialplan) are written as "≺X>5551234", ype and may be either "+" (international), "N" (national), ). written as userinfo@domain.tld, like in the respective SIP	

The following screen shows the second sub rule that converts E.164 for the internal called number into a 4-digit extension:

car Address Map Conlightation	
Address map name:	Outbound
Rule name:	Called - Internal
Called address expression:	^\+1716639(\d(4))
Called address format:	\$1
Calling address expression:	
Calling address format:	
Redirect address expression:	
Redirect address format:	
Stop on match:	
NOTE for call address formats: - Addresses received from PSTN (or where <x> represents the number "6" (subscriber) or empty (unknown - Addresses received from SIP are headers.</x>	rr those normalized via dialplan) are written as "≺X>5551234", bype and may be either "+" (international), "N" (national), ). written as userinfo@domain.tid, like in the respective SIP

cuit Address Map Conliguration		
Address map name:	Outbound	
Rule name:	Called - National	
Called address expression:	^\+1	
Called address format:	91	
Calling address expression:		
Calling address format:		
Redirect address expression:		
Redirect address format:		
Stop on match:		
NOTE for call address formats: - Addresses received from PSTN (or where <x> represents the number b "S" (subscriber) or empty (unknown) - Addresses received from SIP are v headers.</x>	r those normalized via dialplan) are written as "«X»5551234", /pe and may be either "+" (international), "N" (national), written as userinfo@domain.tld, like in the respective SIP	

The following sub rule converts the E.164 national number into a 10-digit national number with prefix 91:

The following example converts international call numbers:

🚰 http://127.0.0.1:10005 - Edit Addres	s Map - Microsoft Internet Explorer	_ 🗆 ×
		A
Edit Address Map Configuratio	n	$\odot$
Address map name:	Outbound	
Rule name:	Called - International	
Called address expression:	<u>`\+</u>	
Called address format:	9011	
Calling address expression:		
Calling address format:		
Redirect address expression:		
Redirect address format:		
Stop on match:		
NOTE for call address formats: - Addresses received from PSTN where <x> represents the numbe "S" (subscriber) or empty (unknov - Addresses received from SIP ar headers.</x>	(or those normalized via dialplan) are written as "≺X>5551234", Ir type and may be either "+" (international), "N" (national), vn). e written as userinfo@domain.tld, like in the respective SIP	
	OK Cancel	
5A -		V
E Done	j j j j 🖉 Internet	1

Once an address map rule is created, it can be applied in three different places. To ease the configuration and troubleshooting processes, apply the rule on the outbound route as shown below:

Routing Configuration					
Name	Sources	Destinations	Address Map	Enabled	
Outbound Route	Mediation Server	Controller1 (Slave), Controller2 (Slave), Controller3 (Slave), Controller4 (Slave)		₹	↑ L Details
Inbound Route	Controller1, Controller2, Controller3, Controller4	Mediation Server (Slave)	none 💌	₹	f L Details
		Add			

## Inbound call example using address map:

This example assumes that the PBX sends inbound calls using a 4-digit extension, with the ISDN type of number flag set to Subscriber for internal numbers, National for national calls, and International for international calls.

Called number	From PBX	To Microsoft <sup>®</sup> OCS
Internal	xxxx (with subscriber type of number)	+1716639xxxx

Calling number	From PBX	To Microsoft <sup>®</sup> OCS
Calling from internal	xxxx (with subscriber type of number)	+1716639xxxx
Calling from national	xxxxxxxxxx (with national type of number)	+1xxxxxxxxx
Calling from international	xxxxxx (with international type of number)	+xxxxxx

Sub rule name	Expression	Format	Stop on match
Called	^S(\d{4})\$	+1716639\$1	Not checked
Calling - internal	^S(\d{4})\$	+1716639\$1	Checked
Calling - national	^N(\d{10})\$	+1\$1	Checked
Calling - international	^\+	+	Checked

ddress Map Configuration				۲
Name	Rule Name	Stop on Match	Enabled	
	Calling number	Г	1	↑ ↓ Details Dele
	Called - Internal	7	<b>V</b>	↑ ↓ Details Dele
Outbound	Called - National	<b>v</b>	V	↑ ↓ Details Dele
	Called - International	<b>V</b>	V	↑ ↓ Details Dele
	Add Rule			
	Called		V	↑ ↓ Details Dele
	Calling - Internal	<b>v</b>	▼	↑ ↓ Details Dele
Inbound	Calling - National	<b>v</b>	V	↑ ↓ Details Dele
	Calling - International	<b>v</b>	V	↑ ↓ Details Dele
	Add Rule			
		Add		

## Create an address map named Inbound and its four sub rules as shown below:

dit Address Map Configuration	
Address map name:	Inbound
Rule name:	Called
Called address expression:	^S(\d(4))\$
Called address format:	+1716639\$1
Calling address expression:	
Calling address format:	
Redirect address expression:	
Redirect address format:	
Stop on match:	
NOTE for call address formats: - Addresses received from PSTN (o where ≺X> represents the number i "S" (subscriber) or empty (unknown - Addresses received from SIP are i headers.	or those normalized via dialplan) are written as "<>>5551234", type and may be either "+" (international), "N" (national), )), written as userinfo@domain.tld, like in the respective SIP
	OK Cancel

#### NEC NEAX2400 IPX

	Map - Microsort Internet Explorer	
dit Address Map Configuration		6
Address map name:	Inbound	
Rule name:	Calling - Internal	
Called address expression:		
Called address format:		
Calling address expression:	^S(\d(4))\$	
Calling address format:	+1716639\$1	
Redirect address expression:		
Redirect address format:		
Stop on match:	V	
NOTE for call address formats: - Addresses received from PSTN (( where <x> represents the number "S" (subscriber) or empty (unknown - Addresses received from SIP are headers.</x>	or those normalized via dialplan) are written as " <x>55512 'type and may be either '*' (international), "N" (national), n), written as userinfo@domain.ttd, like in the respective SIP</x>	34",
NOTE for call address formats: - Addresses received from PSTN (( where <x> represents the number "6" (subscriber) or empty (unknown - Addresses received from SIP are headers.</x>	or those normalized via dialplan) are written as "<>55512 type and may be either "+" (international), "N" (national), n), written as userinfo@domain.tld, like in the respective SIP OK Cancel	34",
NOTE for call address formats: - Addresses received from PSTN (( where <x> represents the number "S" (subscriber) or empty (unknow - Addresses received from SIP are headers.</x>	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), n), written as userinfo@domain.tld, like in the respective SIP OK Cancel</x>	34",
NOTE for call address formats: - Addresses received from PSTN (( where  - represents the number "S" (subscriber) or empty (unknown - Addresses received from SIP are headers. re p;//127.0.0.1:10005 - Edit Address	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), n), written as userinfo@domain.tld, like in the respective SIP OK Cancel OK Cancel Map - Microsoft Internet Explorer</x>	34",
NOTE for call address formats: - Addresses received from PSTN (d where < <pre>x* represents the number '6" (subscriber) or empty (unknown - Addresses received from SIP are headers. ne p://127.0.0.1:10005 - Edit Address</pre>	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), n), written as userinfo@domain.tld, like in the respective SIP OK Cancel Map - Microsoft Internet Explorer</x>	34",
NOTE for call address formats: - Addresses received from PSTN (( where <x> represents the number "S" (subscriber) or empty (unknowu - Addresses received from SIP are headers. p://127.0.0.1:10005 - Edit Address dit Address Map Configuration</x>	or those normalized via dialplan) are written as "«X>55512 Type and may be either "+" (international), "N" (national), n), written as userinfo@domain.ttd, like in the respective SIP OK Cancel Map - Microsoft Internet Explorer	34", 
NOTE for call address formats: - Addresses received from PSTN (( where <x> represents the number "S" (subscriber) or empty (unknown - Addresses received from SIP are headers. he p://127.0.0.1:10005 - Edit Address fold Address Map Configuration Address map name:</x>	or those normalized via dialplan) are written as " <x>55512 type and may be either '*' (international), "N" (national), written as userinfo@domain.ttd, like in the respective SIP OK Cancel Map - Microsoft Internet Explorer</x>	34",
NOTE for call address formats: - Addresses received from PSTN (4 where  - represents the number 'S" (subscriber) or empty (unknown - Addresses received from SIP are headers. 	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), n), written as userinfo@domain.tld, like in the respective SIP OK Cancel OK Cancel Map - Microsoft Internet Explorer</x>	34",
NOTE for call address formats: - Addresses received from PSTN (4 where  - Addresses received from SIP are headers. - Addresses received from SIP are headers. 	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), ), written as userinfo@domain.tld, like in the respective SIP OK Cancel OK Cancel Map - Microsoft Internet Explorer</x>	34",
NOTE for call address formats: - Addresses received from PSTN (4 where <x> represents the number "S" (subscriber) or empty (unknown - Addresses received from SIP are headers. </x>	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), written as userinfo@domain.tld, like in the respective SIP OK Cancel OK Cancel Map - Microsoft Internet Explorer</x>	34",
NOTE for call address formats: - Addresses received from PSTN (d where - represents the number "S" (subscriber) or empty (unknown - Addresses received from SIP are headers. 	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), ), written as userinfo@domain.tld, like in the respective SIP OK Cancel Map - Microsoft Internet Explorer Inbound Calling - National Calling - National</x>	
NOTE for call address formats: - Addresses received from PSTN (d where <x> represents the number 'S' (subscriber) or empty (unknown - Addresses received from SIP are headers. </x>	or those normalized via dialplan) are written as " <x>55512 type and may be either "+" (international), "N" (national), ), written as userinfo@domain.tld, like in the respective SIP OK Cancel Map - Microsoft Internet Explorer Inbound Calling - National Calling - National</x>	34", 

Redirect address format: V Stop on match: NOTE for call address formats: - Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234", where <X> represents the number type and may be either "+" (international), "N" (national), "S" (subscriber) or empty (unknown). - Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP headers. OK Cancel 🙆 Done 📄 📄 🚺 🎯 Internet

Address map name: Rule name: Called address expression: Called address format:	Inbound Calling - International	
Rule name: Called address expression: Called address format:	Calling - International	
Called address expression: Called address format:		
Called address format:		
Calling address expression:	^\+	
Calling address format:	+	
Redirect address expression:		
Redirect address format:		
Stop on match:	<u>र</u>	
NOTE for call address formats: - Addresses received from PSTN (or the where > represents the number type "S" (subscriber) or empty (unknown). - Addresses received from SIP are write - Addresses	nose normalized via dialplan) are written as " <x>5551234", e and may be either "+" (international), "N" (national), tten as userinfo@domain.ttd, like in the respective SIP</x>	

Apply the address map inbound rule on the inbound route as follows:

Routing Configuration					6
Name	Sources	Destinations	Address Map	Enabled	
Outbound Route	Mediation Server	Controller1 (Slave), Controller2 (Slave), Controller3 (Slave), Controller4 (Slave)	Outbound 💌	1	† L Details
Inbound Route	Controller1, Controller2, Controller3, Controller4	Mediation Server (Slave)	Inbound V	5	1 L Details
		Add			

## 5.2.5 Restarting the Dialogic<sup>®</sup> Diva<sup>®</sup> SIPcontrol<sup>™</sup> Software

Note: A restart of the Diva SIPcontrol software service is needed only if the setting under Network Interface is changed.

Save the configuration and restart the Diva SIPcontrol software service for the changes to take effect. To do so, click Service Status on the left hand side of the main configuration page, and then click Restart SIPcontrol. The Last operation log will show that the service has been stopped and started again.

NEC NEAX2400 IPX

## 6. PBX Setup Notes

The basic steps of setting up the PBX for use with this Dialogic<sup>®</sup> 4000 Media Gateway Series (DMG4000 Gateway) and a voice processing system are as follows:

- 1. Initializing the ISDN services.
- 2. Enabling ANI pass-through.
- 3. Building and configuring trunk routing.
- 4. Setting up the subscriber station sets.

All PBX programming is done via a Windows<sup>®</sup>-based application that connects to the serial administration port of the PBX.

## 6.1 Initializing the ISDN Services

Using the ASYD (Assignment of System Data), set the system data bytes that are required to bring the ISDN system into service.

Note that not all fields will be visible at the outset. As you start to enter data, however, new fields will appear.

🎄 ASYD - NEXT-P	вх	
<u>File V</u> iew <u>H</u> elp		
ASYD(Assignm	ent of System Data)	
SYS		
1		
INDEX	DATA	
186	63	
FDATA	MDATA	
00	00	

In the ASYD - Next-PBX dialog box, configure the following:

- In the SYS field, enter 1.
- In the INDEX field, enter 186.
- In the DATA field, enter 63.
- Press [Enter] to save and move on to Index 187.
- In the ASYD dialog box in the DATA field, enter 00.
- Press [Enter] to save Index 187.
- Close the ASYD Next-PBX dialog box.

## 6.2 Enabling ANI Pass-Through

Use the  ${\tt ASFC}$  (Assignment of Service Feature Restriction Class) command to allow ANI to pass from the PBX to TELCO.

Note that *SFI* 94 must be set to 0. For extensions that want outward ANI to be blocked, assign them to an *SFC* that has *SFI* 94 set to a 1. Always assign *SFIs* 97 and 98 as 0.

🛃 ASFC - NEXT-PBX	
<u>Fi</u> le <u>V</u> iew <u>H</u> elp	
ASFC(Assignment of Service Feature Restriction Class)	
D/N TN SFI D 1 94	
SFC 0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15	
RES 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0	

In the <code>ASFC - Next-PBX</code> dialog box, configure the following:

- In the D/N field, enter D (day time configuration).
- In the TN field, enter 1.
- In the SFI field, enter 94.
- In the RES fields, enter 0 for every field.
- Close the ASFC Next-PBX dialog box.

## 6.3 Building and Configuring Trunk Routing

**Step 1:** Use the ARTD (Assign Rout Data) command to assign a route data for each B-channel and D-channel route.

For B-channels, setting CDN to 1 allows INCOMING ANI to D-term displays (does not affect analog stations). This does NOT affect outgoing ANI. Other CDNs may be left at their default values (data 0) for the route.

🛃 A	RTD - N	IEXT-PBX			
Eile	⊻iew <u>F</u>	<u>t</u> elp			
AR'	TD (Assig	a)			
	RT 5				<u>G</u> et
	- Quick	Edit		1	
	CDN	FUNC	DATA		<u>S</u> et
					<u></u> el
	CDN	FUNC	DATA	^	
	1	OSGS	6		Exit
	2	ONSG	0		
	3	ISGS	0		
	4	INSG	0		
	5	TF	0		
	6	TCL	1		
	7	LT	1		
	8	RLP	0		
	9	TQ	0		
	10	SMDR	0		
	11	TD	0		
	12	DR	0		
	13	AC	1		
	14	TNT	0		
	15	LSG	0		
	16	SMDR2	0		
	17	H/M	0		
	18	MC	0		
	19	ANI	0		
	20	D	0		
	21	MSB	0		
	22	I MSW	1 0		

In the ARTD - Next-PBX dialog box, configure the following:

- In the RT field, enter a number from 1-255. Note: 31 is the unofficial convention. Also, remember this route number for later steps in the configuration.
- Click the GET button to initiate the command.

- Enter 1 in the following fields:
  - o TCL
  - o LT
  - o AC
- Click the SET button to initiate the command.
- Close the ARTD Next-PBX dialog box.

**Step 2:** Use the ARTI (Assignment of Trunk Application Data) command to configure your Q.SIG line for the ISO standard protocol.

🛃 ART	11 - N	EXT-PBX			
<u>File V</u> ie	ew <u>F</u>	<u>t</u> elp			
ABTL	Assiar	nment of Trunk.	Application	n Data)	
				,	
RI	r  10	<u>G</u> et			
- C	Juick I	Edit		_	
	CDN	Set			
					Del
	DN	FUNC	DATA		
$\frac{4}{4}$	1		U 0		E <u>x</u> it
4	2	ECCISTD	U 0		
4	3	MFCG2	U 0		
4	4		U 0		
4	5		U 0		
4	5	VRD	U		
4	/	INID	1		
4	8	JECUIS	U U		
4	9	ECCIS2	0		
5	U	IPINI	0		
5	1	IPTRK	0		
5	2	CTCF	1		
5	3	HERT	1		
5	4	DCANS	0		
5	5	RND	0		
5	6	CLBK	0		
5	7	UALAW	0		
5	8	MCTFAC	0		
5	9	RE	0		
6	0	PR	1		
6	1	СОТ	0		
le:	2	1997	I 0		

In the ARTI - Next-PBX dialog box, configure the following:

- In RT field, enter a B-channel route number within the range of 1 through 255.
- Click the GET button to initiate the command.
- Enter 1 in the following fields:

- O INTD
- o CTCFo RERT
- O REI O PR
- Click the SET button to initiate the command.
- Close the ARTI Next-PBX dialog box.

Other CDNs may be left at their default values (data 0) for the route.

**Step 3:** Use the ATRK (Assignment of Trunk Data) command to assign trunks to the defined B-channel and D-channel routes.

There should be two trunks programmed for the D-channel route. The trunks assigned will depend on the circuit cards used.

🛃 ATRK - NEXT-PBX	
<u>File V</u> iew <u>H</u> elp	
ATRK(Assignment of Trunk Data)	
RT TK	<u>G</u> et
10  1	<u>S</u> et
LENS TN	Del
002090  1	E <u>x</u> it

In the ATRK - Next-PBX dialog box, configure the following:

- In the RT field, enter the route number assigned in step 1.
- In the TK field, enter a trunk number used for calls to and from the T1.
- In the LENS field, enter the LENS (card address) number of the T1 card. A trunk number used for calls to and from T1.
- In the TN field, enter a tenant number used for calls to and from the T1.
- Click the SET button to initiate the command.
- Close the ATRK Next-PBX dialog box.

Repeat these steps to assign a second trunk (TK) on the T1 card.

**Step 4:** Use the ARSC (Assignment of Route Restriction Class) command to allow, restrict, or toll restrict calls to the B-channel route.

🛃 ARSC - NE	ХТ-РЕ	X									(			×
<u>F</u> ile ⊻iew <u>H</u> el	P													
ABSC (Assignment of Boute Restriction Class)														
					,							<u>G</u> et		
D/N D											-	C -1		1
	_			_							_	<u>5</u> et		
TN 1		RT	100								Exit			
											_			
Dec. 1			-		71		0	101		10	10	1.4	4.51	
BBL0		<u> </u>	4	5 5 1 1		8	9	10	1	1	13	14	15	
BBI-1	1 1	1 1	1	1 1	1	1	1	1	1	1	1	1	1	
RRI-2	1 1	1 1	1	1 1	1	1	1	1	1	1	1	1	1	
RRI-3	1 1	1 1	1	1 1	1	1	1	1	1	1	1	1	1	

In the ARSC - Next-PBX dialog box, configure the following:

- In the D/N field, enter D for the day time configuration.
- In the TN field, enter the tenant number used for calls to and from the T1.
- In the RT field, enter the route number assigned in step 1.
- In the RSC table, enter 1 for all calls.
- Click the SET button to initiate the command.
- Close the ARSC Next-PBX dialog box.

If night time configuration is also required, enter  $\mathbb{N}$  in the  $D/\mathbb{N}$  field and repeat the steps above.

**Step5:** Use the ARRC (Assignment of Alternative Route Restriction) command to assign route to route connection data to allow tandem connections between existing routes and the B-channel route.

💑 ARRC - NEXT-PBX	
<u>File V</u> iew <u>H</u> elp	
ARRC(Assignment of Alternative Route Restriction)	
ICRT OGRT	
ARI A-RES ARI D-RES	
1 1	

In the ARRC - Next-PBX dialog box, configure the following:

- In the ICRT field, enter 1 (incoming route number).
- In the OGRT field, enter 1 (outgoing route number).
- Close the ARRC Next-PBX dialog box.

**Step 6:** Use the ADPC (Assignment of Determinate Point Code Data) command to assign a separate and unique point code for each ISDN span.

The allowed range for point codes is 1 through 16383. Any number in that range can be used; however, it is recommended to use a point code under 200. Also, you cannot use any point code that is already assigned in the PBX, which is used for CCIS or ISDN. To find point codes that are in use, look in AYSD Index 180 & 181 as well as LDPC.

<b>3</b>	DPC -	NE	хт-р	BX									
Eile	⊻iew	Help	2										
	ADP	C(As	sign	men	t of Det	termin	nati	e Po	oint C	ode	Data	)	
	R	т			PC								
	10												

In the ADPC - Next-PBX dialog box, configure the following:

- In the RT field, enter the route number assigned in step 1.
- In the PC field, enter the point code you wish to use.
- Close the ARSC Next-PBX dialog box.

**Step 7:** Use the ACSC (Assignment of CSC Data) command to assign only an even CSCG number. If ACSC data must be deleted, first delete ACIC/ACIC1 data, then delete ACSC.

🛃 ACSC	- NEXT-PE	ЗХ		
<u>F</u> ile ⊻iev	v <u>H</u> elp			
ACSO	C(Assignm	ent of CSC Data	a)	
	CSCG			
	130			
	GROUP	ССН		
	0	00112		
	1			
	2			
	3			

In the ACSC - Next-PBX dialog box, configure the following:

- In the CSCG field, enter an available common channel signaling controller group number from 2 through 255. Do not use 128 or 129.
- In the CCH field, enter the slot address (MG, UNIT, and GROUP).
- Close the ARSC Next-PBX dialog box.

Step 8: Use the <code>ACIC1 (Assignment of CIC Code Data 1)</code> command on <code>IMX/IPX PBX's using Windows<sup>®</sup> commands (ACIC1 / ACIC2)</code>.

🗸 ACIC1 - NEXT-PBX	
File View Help	
ACIC1 (Assignment of CIC Code Data 1)	
PC CSCG	

In the ACIC1 - Next-PBX dialog box, configure the following:

- In the PC field, enter the point code for the ISDN span that was assigned in step 6.
- In the CSCG field, enter an even number that was assigned in step 7.
- Close the ACIC1 Next-PBX dialog box.

Step 9: Use the <code>ACIC2 (Assignment of CIC Code Data 2)</code> command on <code>IMX/IPX PBX's using Windows®</code> commands (ACIC1 / ACIC2)

🛃 ACIC2 - NEXT-PBX	
<u>File V</u> iew <u>H</u> elp	
ACIC2(Assignment of CIC Code Data 2)	
PC CIC	

In the ACIC2 - Next-PBX dialog box, configure the following:

- In the PC field, enter a point code for the ISDN span that was assigned in step 6.
- In the CIC field, enter 1.
- In the LENS field, enter the first B-channel trunk assigned in step 3.
- Close the ACIC2 Next-PBX dialog box.

Note: Repeat the ACIC2 steps for every circuit identification code (CIC) and its corresponding line equipment number (LENS). Make sure you only assign a CIC to a B-channel trunk.

**Step 10:** Use the ASPA (Assignment of Special Access Code) command on IMX/IPX PBXs to create the access code that is going to be used to route over the trunk.

🛃 ASPA - IPX_	Р		
<u>F</u> ile ⊻iew <u>H</u> elp			
ASPA(Assign	ment of Special Acce	ess Code)	
TN	ACC	CI	
1	80	N	

In the ASPA - IPX\_IP dialog box, configure the following:

- In the TN field enter the TN of the PBX.
- In ACC field, enter the access code you want to use to dial across the trunk.
- In the CI field, enter the hooking state. In this example, normal hooking was used.

**Step 11:** Use the AFRS (Assignment of Flexible Route Selection) command on IMX/IPX PBXs to tie the access code to a route.

👼 AFRS - IPX_IP			
<u>F</u> ile ⊻iew <u>H</u> elp			
AFRS(Assignn	nent of Flexib	le Route Selection)	
TN	RT	NPC	
1	31	80	

In the AFRS - IPX\_IP dialog box, configure the following:

- In the TN field enter the TN of the PBX.
- In RT field, enter the "dummy" route used to connect to the trunk route.
- In the NPC field, enter the access code created in the previous step.

**Step 12:** Use the AOPR (Assignment of Outgoing Pattern Routing Data) command on IMX/IPX PBXs to tie the access code to a trunk route.

AOPR - IPX_IP		
<u>File V</u> iew <u>H</u> elp		
AOPR(Assignme	ent of Outgoing I	Pattern Routing Data)
TDPTN	0PR	RA
0	80	0
Е	RT	SKIP
0	100	2
PNL	0VFT	PRSC
0	0	0
		,
DEL?	_	

In the AOPR - IPX\_IP dialog box, configure the following:

- In the TDPTN field, enter the time of day pattern on the PBX.
- In the OPR field, enter the access code created in the previous steps.
- In RA field, enter the route advancing order.
- In the E field, enter the route advancing end indication.
- In RT field, enter the trunk route used to connect to the trunk.
- In the SKIP field, enter the number of digits to be skipped.
- In the PNL field, enter the pattern number location.
- In the OVFT field, enter the overflow tone operator, where 0 = on and 1 = off.
- In the PRSC field, enter the priority restriction class.

Step 13: Use the MBTK (Make Busy of Trunk) command to set the B-channels into an idle state.

🚑 МВТК - NEXT-PBX	
<u>File V</u> iew <u>H</u> elp	
MBTK(Make Busy of Trunk)	
RT TK	<u>G</u> et
	<u>S</u> et
мв	<u>C</u> lose

In the MBTK - Next-PBX dialog box, configure the following:

- In RT field, enter the route number entered in step 1.
- In TK field, enter the trunk number entered in step 3.
- In MB field, enter 0, where 0 sets the trunk to idle on the T1.
- Click SET, to initiate the command.
- Close the MBTK Next-PBX dialog box.

Repeat this step for each for each trunk used.

ABTC - NEXT-PBX <u>File View H</u>elp MBTC (Make Busy of Trunk - Continuous) <u>S</u>et START END E<u>x</u>it BT TK MB RT TK MB STATUS < >

Step 14: Use the MBTC (Make Busy of Trunk - Continuous) command to set all the B-channels to idle.

In the MBTC - Next-PBX dialog box, configure the following:

- In the RT field, enter the route number entered in step 1.
- In the TK START field, enter the first trunk number entered in step 3.
- In the TK END field, enter the last trunk number entered in step 3
- In the MB field, enter 0, where 0 sets the trunk to idle on T1.
- In the MBTC window hit SET, to initiate the command.
- Close the MBTC Next-PBX dialog box.

Step 15: Use the MBRT (Make Busy or Route) command to set the B-channels idle.

🛃 MBRT -	NEXT-PBX	
<u>F</u> ile ⊻iew	<u>H</u> elp	
MBRT(Mał RT	ke Busy of Route) MB ☞ Make Idle ⓒ Make Busy	<u>S</u> et E <u>x</u> it

In the MBRT - Next-PBX dialog box, configure the following:

- In the RT field, enter the route number entered in step 1.
- In the MB field, enable Make Idle to set the trunk to idle on T1.
- Close the MBRT Next-PBX dialog box.

**Step 16:** Verify that the D-channel is stable. The connecting PBX should see the D-channel linked, and, at this time, the B-channels should come online. This is sometimes referred to as a B-channel restart. Test calls can be made on incoming routes.

**Step 17:** Set up to Least Cost Routing (LCR) rules. It is suggested that you see the NEAX 2400 ISDN Installation manual or contact local NEC vendor representative for proper configuration, as many of these parameters are site specific.

**Step 18:** Once you have completed the LCR setup, the next step is time to program ANI "Caller ID" to assign caller ID data for outgoing calls. See the NEC2400 ISDN Installation manual, or contact local NEC vendor representative for proper configuration, as many parameters are site specific.

## 7. Microsoft<sup>®</sup> Office Communications Server 2007 (OCS) Setup

## 7.1 Steps for configuring Microsoft<sup>®</sup> OCS

Normalization rules are used to convert dial numbers into full E.164 formatted numbers. Microsoft<sup>®</sup> OCS uses the standard E.164 format to search for users listed in the Active Directory (AD).

If a Microsoft<sup>®</sup> OCS user dials an internal extension number (normally 3-5 digits), the normalization rules convert it into full E.164 format. These normalization rules should cover dialed digits for internal extensions, local numbers, long distance numbers, and international numbers.

To configure Microsoft<sup>®</sup> OCS, click Start > Programs > Administrative Tools > OCS 2007.

On the tree presented in the configuration window, right-click Forest then select Properties and then Voice Properties form the menu provided. Edit a location profile as shown in the following example:

Location	Profile				
<u>N</u> ame:		Location <sup>*</sup>	1		
<u>D</u> escription		,			
test					A V
Normalizatio	n Rules ——				
Normalizatio	on rules are pro	cessed in list o	rder; plea	ase use the b	uttons on the
side for adju	usting the order.				
5xxx					[
5xxx 3xxx 2xxx					
5xxx 3xxx 2xxx 4 digits					<u>U</u> p
5xxx 3xxx 2xxx 4 digits					Шр
5xxx 3xxx 2xxx 4 digits					<u>Up</u> D <u>o</u> wn
5xxx 3xxx 2xxx 4 digits					Up D <u>o</u> wn
5xxx 3xxx 2xxx 4 digits					Up D <u>o</u> wn
5xxx 3xxx 2xxx 4 digits	<u>A</u> dd	<u>E</u> dit		<u>R</u> emove	Up D <u>o</u> wn

Click Add or Edit to create or change a particular rule.

lit Phone Number No	rmalization Rule		
<u>N</u> ame:	4 digits		
Click to copy an existing	) rule.		<u>С</u> ору
Description:			
any 4 digits			<u></u>
			-
Translation			
Phone pattern regula	expression:		
^(\d{4})\$			
<u>T</u> ranslation pattern re	gular expression:		
+1716639\$1			
Click Helper for assist regular expressions a	ance in creating con nd translations.	nmon phone number	Helper
Test translation			
To test the translation pattern, the translatio	, enter a sample dial n will be shown.	ed number. If it matcl	hes the phone
Sample dialed numbe	r:		
, Translated number:			

In this example, when a user dials any 4-digit number, it will be converted to its E.164 equivalent of +1716639xxxx and then that number will be searched for in AD.

Name	Phone Pattern	Translation Pattern	Comments
Extensions	^(\d{4})\$	+1716639\$1	Internal extensions
Local	^(\d{7})\$	+1716\$1	Local number
National	^1(\d*)\$	+1\$1	Long distance number
International	^011(\d*)	+\$1	International number

More examples are shown in the following table:

A default route is used to route all calls to Microsoft<sup>®</sup> Mediation Server. If you need to route some calls to a different Microsoft<sup>®</sup> Mediation Server, configure the Target phone numbers field accordingly.

To configure Microsoft<sup>®</sup> OCS, click Start > Programs > Administrative Tools > OCS 2007.

On the tree presented in the configuration window, right-click Forest then select Properties and then Voice Properties form the menu provided. Edit a route as shown in the example below.

dit Route			
<u>N</u> ame:	Universal Route		
Description:			
Routes every call			*
			Ŧ
A route requires a tar	get phone number regu r more phone usages	llar expressior	), one or more
gateways, and one o	r more priorie usages.		
<ul> <li>Target phone numb</li> </ul>	pers:		
<u>T</u> arget regular expr	ession		
^\+?(\d*)\$			
			Helper
dmg4000.Buf0CS	6.local:5061		
1		<u>лл</u> 1	Demons
		<u>A</u> aa	<u>H</u> emove
Phone usages			
Default Usage			
1			
			<u>C</u> onfigure

This entry routes numbers with or without "+" prefix followed by any digits to Microsoft<sup>®</sup> Mediation Server dmg4000.bufocs.local.

Restart the Front End Services for the above changes to take effect, including all normalization rules. This can be done from the window Services.

Note: Unless the dialed number from Microsoft<sup>®</sup> OCS client (such as Microsoft<sup>®</sup> Office Communicator) is in E.164 format, Microsoft<sup>®</sup> OCS must find a normalization rule to convert the dialed number to E.164. If no rule is found

or matched, outbound calls will fail. In this case, Dialogic<sup>®</sup> Diva<sup>®</sup> Diagnostics trace will not receive an outbound SIP message, since the call will not yet have reached the SIP gateway.

# 7.2 Steps for configuring Microsoft<sup>®</sup> Office Communications Server 2007 (OCS) clients

🚡 Console1 - [Console Root\Active Di	rectory Users and	Computers [ocste	est01.BufOCS.local]\BufOCS	.local\Users]
🚡 Eile Action View Favorites Wir	ndow <u>H</u> elp			
	a 🗈 😰 🖬	🧞 👯 🐂 🖓 🤞	s 🗑	
	Name	Tupe	Description	1
Microsoft Office Communications Se		User	Built-in account for admini	
Forest - BufOCS.local		User	balle in account for damining	
🗄 📄 Enterprise pools		Security Group	Members of this group are	
🖃 💼 Standard Edition Servers	CERTSVC D	Jocani, arcap in		
🖻 🚺 ocstest01	🖸 Dave Bergguist	Ray Cassick Prop	erties	? ×
😟 🖳 🛄 Users		Member Of	Dial-in Enviror	oment Sessions
	DHCP Users	General 1 Add	ress   Account   Profile   1	
	<b>DnsAdmins</b>	Remote control	Torminal Convines Profile	COM. Communications
Onassigned users	DnsUpdatePr			
Heulation Servers	Domain Admins	Enable use	er for Office Communications Ser	ver
Event Viewer (Local)	🚮 Domain Com	Sign-in name:		
Services (Local)	🕵 Domain Cont	sintreassi	ck a Buildes	
🖃 🔏 Active Directory Users and Comput	🕵 Domain Guests	[sip.redssi		
🗄 💼 Saved Queries	💯 Domain Users	Server or pool:	:	
🖻 🗊 BufOCS.local	Enterprise A	ocstest01	I.BufOCS.local	▼
🗄 🛄 Builtin	Group Policy	- Meetings-		
E Computers	5 Guest	Allow an	nonvmous participants	
H	HelpServices			
	WPG	Policy:	Default Policy	
Users	IUSR_OCSTE			View
	IWAM_OCST			
	🖸 John Doe	Note: Meeti allows per u	ng settings cannot be changed i iser configuration	unless the global setting
	RAS and IAS	allows per a	ser conligaration.	
	Ray Cassick			
		Additional opti		Configure
	RTC Local Ad	Auditional opti	uris.	
	RTC Server			
	RTC Server L			
	RTCArchivina		OK Cancel	Apply Help
	RTCCompon			
	RTCCompon	Security Group	Members can be used as	
	TCHSUniver	Security Group	Members can be used as	
	🕵 RTCProxyUni	Security Group	Members can be used as	
	RTCService	User	Service account of Office	
	🕼 RTCUniversal	Security Group	Members have read acces	

The domain users need to be enabled for making calls through Microsoft<sup>®</sup> OCS.

Under the Communications tab, check the Enable user for Office Communications Server option and then click the Configure button.



In the above configuration for the hypothetical user Ray Cassick, an inbound PSTN call for 3101 will be converted by the Dialogic<sup>®</sup> Diva<sup>®</sup> SIPcontrol<sup>™</sup> Software to +17166393101 because in the Diva SIPcontrol software dialpan in the SIP Peer Configuration section under Address Normalization the:

- Number format (outbound) is set to International number, and
- Encoding (outbound) is set to Use type flag.

Microsoft<sup>®</sup> OCS will ring the user Ray Cassick if he is logged on to Microsoft<sup>®</sup> OCS from Microsoft<sup>®</sup> Office Communicator or any Microsoft<sup>®</sup> OCS supported device.

## 8. Microsoft<sup>®</sup> Mediation Server Installation and Configuration

## 8.1 Installation

The gateways of the Dialogic<sup>®</sup> 4000 Media Gateway Series (DMG4000 Gateways) are shipped with pre-installed Microsoft<sup>®</sup> Mediation Server software. You can complete the Microsoft<sup>®</sup> Mediation Server configuration by running Microsoft<sup>®</sup> Office Communications Server 2007 (OCS) "Setup.exe" in the DMG4000 Gateways. In the Microsoft<sup>®</sup> OCS Deployment Wizard, select Deploy Other Server Roles, then select Deploy Mediation Sever. Follow the steps in the Wizard to complete the setup:

**Step 1**: Install the Microsoft<sup>®</sup> Mediation Server software.

**Step 2**: Activate Microsoft<sup>®</sup> Mediation Server. Use the existing account and enter the password for the service account.

**Step 3**: No action needed. Do this step when the installation is complete.

### Step 4: Configure Certificate.

- 1. Download the CA certification path for Microsoft<sup>®</sup> Mediation Server.
  - From Start > Run, enter <u>http://<CA server>/certsrv</u>
  - Select to download a CA certificate, chain or CRL.
  - Click Download CA certificate chain.
  - In File Download, click Save.
- 2. Install the certificate chain for the Microsoft<sup>®</sup> Mediation Server:
  - In the Deployment Wizard, run step 4 again.
  - Select Import a certificate chain from a .p7b file in step 1.
- 3. Verify that your CA is in the list of Trusted root CAs:
  - In the Microsoft<sup>®</sup> Management Console (MMC) snap-in, click Certificates (If not already done, add it.)
  - Verify that CA is on the list of trusted CAs as shown in the example below.

🚡 Console1 - [Console Root\Certificates (Local Computer)\Trusted Root Certification Authorities\Certificates]					
📸 Eile Action View Favorites Window !	<u>H</u> elp				
		[- ·-	( <u> </u>	[	_
Console Root		Issued By	Expiration Date		Ē
Services (Local)	Equirax Secure Certificate Authority	Equirax Secure Certificate Authority	8/22/2018	Secure Email, Server	E
- C Diva Server Management		Equirax Secure eBusiness CA-1	6/20/2020	Secure Email, Server	E
		Equirax Secure eBusiness CA-2	6/23/2019	Secure Email, Server	E
Active Connections	Equitax Secure Global eBusiness C	Equitax Secure Global eBusiness CA-1	6/20/2020	Secure Email, Server	E
Call History	EUnet International Root CA	EUnet International Root CA	10/2/2018	Secure Email, Server	E
🗄 🎻 Active Directory Users and Computers [ocst	FESTE, Public Notary Certs	FESTE, Public Notary Certs	1/1/2020	Secure Email, Server	F
🖻 👸 Certificates (Local Computer)	ESTE, Verified Certs	FESTE, Verified Certs	1/1/2020	Secure Email, Server	F
	First Data Digital Certificates Inc	First Data Digital Certificates Inc. Ce	7/3/2019	Server Authenticatio	F
🖃 📄 Trusted Root Certification Authorities	E FNMT Clase 2 CA	FNMT Clase 2 CA	3/18/2019	Secure Email, Server	F
	GlobalSign Root CA	GlobalSign Root CA	1/28/2014	Secure Email, Server	G
🕀 💼 Enterprise Trust	GTE CyberTrust Global Root	GTE CyberTrust Global Root	8/13/2018	Secure Email, Client	G
🗄 💼 Intermediate Certification Authorities	GTE CyberTrust Root	GTE CyberTrust Root	4/3/2004	Secure Email, Client	ē
🕀 💼 Trusted Publishers	GTE CyberTrust Root	GTE CyberTrust Root	2/23/2006	Secure Email, Client	e
😟 🛄 Untrusted Certificates	http://www.valicert.com/	http://www.valicert.com/	6/25/2019	Secure Email, Server	۷
Third-Party Root Certification Authoritie:	http://www.valicert.com/	http://www.valicert.com/	6/25/2019	Secure Email, Server	۷
Trusted People	http://www.valicert.com/	http://www.valicert.com/	6/25/2019	Secure Email, Server	۷
⊕	IPS SERVIDORES	IPS SERVIDORES	12/29/2009	Secure Email, Server	I
	Microsoft Authenticode(tm) Root	Microsoft Authenticode(tm) Root Au	12/31/1999	Secure Email, Code S	Μ
	🔛 Microsoft Root Authority	Microsoft Root Authority	12/31/2020	<a  ></a  >	ľ
	🕮 Microsoft Root Certificate Authority	Microsoft Root Certificate Authority	5/9/2021	<al ></al >	P
	🔛 NetLock Expressz (Class C) Tanusi	NetLock Expressz (Class C) Tanusitv	2/20/2019	Server Authenticatio	Α
	🔛 NetLock Kozjegyzoi (Class A) Tan	NetLock Kozjegyzoi (Class A) Tanusit	2/19/2019	Server Authenticatio	Ν
	🔛 NetLock Uzleti (Class B) Tanusitva	NetLock Uzleti (Class B) Tanusitvany	2/20/2019	Server Authenticatio	Λ
	🔛 NO LIABILITY ACCEPTED, (c)97 V	NO LIABILITY ACCEPTED, (c)97 Veri	1/7/2004	Time Stamping	V
	CSTest	OCSTest	9/7/2012	<all></all>	<
	🔤 PTT Post Root CA	PTT Post Root CA	6/26/2019	Secure Email, Server	К
	🔤 Saunalahden Serveri CA	Saunalahden Serveri CA	6/25/2019	Secure Email, Server	S
	🔛 Saunalahden Serveri CA	Saunalahden Serveri CA	6/25/2019	Secure Email, Server	S
	Secure Server Certification Autho	Secure Server Certification Authority	1/7/2010	Server Authentication	٧
	SecureNet CA Class A	SecureNet CA Class A	10/16/2009	Secure Email, Server	S
	🔤 SecureNet CA Class B	SecureNet CA Class B	10/16/2009	Secure Email, Server	S
	🔛 SecureNet CA Root	SecureNet CA Root	10/16/2010	Secure Email, Server	s
	🖼 SecureNet CA SGC Root	SecureNet CA SGC Root	10/16/2009	Secure Email, Server	s
	🔛 SecureSign RootCA1	SecureSign RootCA1	9/15/2020	Secure Email, Server	J
	SecureSign RootCA2	SecureSign RootCA2	9/15/2020	Secure Email, Server	J
	•				
Trusted Root Certification Authorities store contains 1	04 certificates.				

- 4. Create the certificate request for the Microsoft<sup>®</sup> Mediation Server:
  - Run Deployment Wizard, click step 4.
  - Select the option Create a new certificate.
  - Select the option Send the request immediately to an online CA.
  - Complete the settings in the blank.
  - Click Assign to complete the task.

Note: If you receive the error message "certificate expired or is not yet valid" when you click the assign button at the end of step 4, check the time/time zone configured for your Microsoft<sup>®</sup> Mediation Server is correct, then run the Deployment Wizard again or click Certificates in Available tasks in Microsoft<sup>®</sup> Mediation Server MMC snap-in.

## 8.2 Configuration

From the MMC snap-in, right-click the detected Microsoft<sup>®</sup> Mediation Server and select Properties.



Configure the following settings on the  ${\tt General}$  tab:

dmg4000.bufocs.local Properties					
General Next Hop Connections Certificate	1				
Mediation Server					
EQDN: dmg4000.BufOCS.local					
Communications Server listening IP address:					
192.168.0.106					
<u>G</u> ateway listening IP address:					
192.168.0.106					
A/V Edge Server:					
(None)					
Default location profile:					
Location1 View					
Media port range: 60000 to 64000					
OK Cancel Apply Help					

Click the Next Hop Connections tab and configure the following:

dmg4000.bufocs.local Properties			
General	Next Hop Connections Certificate	,	
Offic Spec PSTN	e Communications Server next hop ify the Office Communications Server used for routing inbound V calls.		
EQDI	N: test01.BufOCS.local		
Port:	: 5061		
PSTN	N Gateway next hop	L	
Specify the PSTN gateway connected to this server.			
<u>I</u> P ad	ddress: 192 . 168 . 0 . 106		
P <u>o</u> rt:	9803		
	OK Cancel Apply Help		

The Port entry under PSTN Gateway Next hop has to match the configuration in the Dialogic<sup>®</sup> Diva<sup>®</sup> SIPcontrol<sup>™</sup> Software under Network Interface Configuration > SIP Listen Port.

Click the Certificate tab.

dmg4000.bufocs.local Properties	×				
General Next Hop Connections Certificate					
Server Certificate					
connections on this server.					
Issued to: dmg4000.BufOC5.local Issued by: OC5Test Valid from 9/11/2007 3:54 PM to 9/10/2009 3:54 PM.					
Select <u>C</u> ertificate Delete Cer <u>t</u> ificate					
Warning: Changing the certificate will have no effect on existing connections. Existing connections will continue to use the old certificate.					
OK Cancel <u>Apply</u> Help					

Select the certificate that will be used to communicate with Microsoft<sup>®</sup> OCS. Microsoft<sup>®</sup> Mediation Server will need to restart for these changes to properly take effect.

## 9. Testing the Validation Matrix

The table below shows various test scenarios that are run as typical validation scenarios if the Dialogic<sup>®</sup> Media Gateway is used in a voice messaging situation. The notes column specifies any notable parts of the test.

The test scenarios below assume that all gateway configuration parameters are at their default values. For a sample showing call flows and states please consult the Gateway SIP Compatibility Guide.

Test Number	Call Scenario Description	Notes
Inbound ca	Il scenarios	
1	Direct call from TDM station set to Microsoft <sup>®</sup> OCS client.	
2	Direct call from Microsoft <sup>®</sup> OCS client to TDM station set.	

## 10. Troubleshooting

## **10.1 Important Debugging Tools**

- Ethereal/Wireshark: Can be used to view and analyze the network captures provided by the Dialogic gateway diagnostic firmware.
- Adobe Audition: Can be used to review and analyze the audio extracted from the network captures to troubleshoot any audio related issues.
- Dialogic<sup>®</sup> Diva<sup>®</sup> Diagnostics tool: Used to review and analyze all SIP and ISDN traffic that relates to calls going into and leaving the Dialogic<sup>®</sup> 4000 Media Gateway.

## 10.2 Using the Dialogic<sup>®</sup> Diva<sup>®</sup> Diagnostics Tool

Before using the Dialogic<sup>®</sup> Diva<sup>®</sup> Diagnostics tool, you would need to enable it by setting the Dialogic<sup>®</sup> Diva<sup>®</sup> SIPcontrol<sup>™</sup> Software debug. To do so, open the Diva SIPcontrol software web interface, click the link System Settings, and set Debug Level to Extended. Click the Save button for the changes to take effect.

Now, you can start the Diva Diagnostics tool. To do so, click: Start > Programs > Dialogic Diva > Diagnostics.

Nova Server Diagnostics						
Eile Edit View Iracing Help						
🔚 👺 🗑 🦻 帐 🛛 🏧 🗈 🖻 🗉 🖌 🔺 🍨 🔳 🏵 🎼 D						
Diva Tracing	Category	ID	Description			
E Trace Masks	D-Channel	D	Displays D-channel data as raw hex frames.			
	Layer 1	1	Displays layer 1 state changes.			
Maintenance Driver	Call Comments (SIG)	С	Displays decoded information for signaling frames and states.			
BAC (WAN Misingert)	Low Layer	0	Displays decoded low layer information			
(WAN Miniport)	Network Layer	N	Displays B-channel establishment and data indication.			
Dive Server Adenters	Data Link Error	F	Displays layer 2 link errors.			
4PRT/F1/T1 #1223 Line 1	Miscellaneous	S	Textual information on call states etc.			
### 4PRI/F1/T1 #1223 Line 2	Extended	S	Displays module-specific detailed information.			
4PRI/E1/T1 #1223 Line 3	B-Channel Data	В	Displays B-channel data as raw hex frames.			
4PRI/E1/T1 #1223 Line 4	Modulation	М	Information on modulation and demodulation.			

- 1. Click one line of your Dialogic<sup>®</sup> Diva<sup>®</sup> Media Board in the left pane and click **B** on the toolbar to activate the Basic tracing level. This level captures Q.931 ISDN messages.
- 2. Click CAPI driver in the left pane and activate the Basic tracing level as explained in step 1.
- 3. Start tracing. To do so, click the start icon 🔶 or select the Start Tracing option form the Tracing menu.
- 4. Reproduce the issue.
- 5. To stop tracing, click the stop icon I on the tool bar or select the Stop Tracing option form the Tracing menu.
- 6. To view your collected trace, click the view icon so on the toolbar or select the View Recorded Trace option from the View menu. A notepad window will open with the recorded log.

# Examples of Dialogic<sup>®</sup> Diva<sup>®</sup> Diagnostics traces for an inbound (TDM to IP) call to Microsoft<sup>®</sup> Office Communications Server 2007 (OCS)

Basic notations for reading the trace:

- SIG-R: RX Q.931 ISDN message
- SIG-X: TX Q.931 ISDN message
- SIPR: RX SIP message
- SIPX: TX SIP message

#### < Below is a RX Q.931 ISDN message for an inbound call >

```
9:16:28.431 C 3 21:2389:383 - SIG-R(030) 08 02 00 17 05 04 03 80 90 A2 18 03 A9 83 81 6C 06 01 A0 33
30 30 32 70 05 C1 35 31 30 31
Q.931 CR0017 SETUP
Bearer Capability 80 90 a2
Channel Id a9 83 81
Calling Party Number 01 a0 '2401'
Called Party Number c1 '5101'
```

#### <Below is a TX SIP message with SDP>

```
9:16:28.431 1 L 12 00010000-SIPX begin to IP:192.168.0.106 port:5060 socket:3 Proto:TCP
9:16:28.431 1 L 12 00010000-
                                 >INVITE sip:+17166395101@dmg4000.bufocs.local:5060 SIP/2.0
9:16:28.431 1 L 12 00010000-
                                 >Via: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
9:16:28.431 1 L 12 00010000-
                                 >Max-Forwards: 70
9:16:28.431 1 L 12 00010000-
                                 >Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, NOTIFY, REFER
9:16:28.431 1 L 12 00010000-
                                 >Accept: application/sdp,application/simple-message-summary
9:16:28.431 1 L 12 00010000-
                                 >Supported: timer, replaces
9:16:28.431 1 L 12 00010000-
                                 >From: "Dialogic Diva SIPcontrol"
<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
9:16:28.431 1 L 12 00010000-
                                 >To: "Default" <sip:+17166395101@bufocs.local;user=phone>
                                 >Call-ID: 9c046698-730448-17@dmg4000
9:16:28.431 1 L 12 00010000-
9:16:28.431 1 L 12 00010000-
                                 >CSeq: 1 INVITE
9:16:28.431 1 L 12 00010000-
                                 >Min-SE: 90
9:16:28.431 1 L 12 00010000-
                                 >Session-Expires: 600;refresher=uac
9:16:28.431 1 L 12 00010000-
                                 >Contact: <sip:+17166392401@192.168.0.106:9803>
9:16:28.431 1 L 12 00010000-
                                 >Content-Type: application/sdp
9:16:28.431 1 L 12 00010000-
                                 >Content-Length: 253
9:16:28.431 1 L 12 00010000-
9:16:28.431 1 L 12 00010000-
                                 >v=0
9:16:28.431 1 L 12 00010000-
                                 >o=SIPcontrol 7472200 7472200 IN IP4 192.168.0.106
9:16:28.431 1 L 12 00010000-
                                 >s=-
9:16:28.431 1 L 12 00010000-
                                 >c=IN IP4 192.168.0.106
9:16:28.431 1 L 12 00010000-
                                 > t = 0 0
9:16:28.431 1 L 12 00010000-
                                 >m=audio 30060 RTP/AVP 8 0 101 13
9:16:28.431 1 L 12 00010000-
                                 >a=rtpmap:8 PCMA/8000
9:16:28.431 1 L 12 00010000-
                                 >a=rtpmap:0 PCMU/8000
9:16:28.431 1 L 12 00010000-
                                 >a=rtpmap:101 telephone-event/8000
9:16:28.431 1 L 12 00010000-
                                 >a=fmtp:101 0-15
9:16:28.431 1 L 12 00010000-
                                 >a=rtpmap:13 CN/8000
                                >a=sendrecv
9:16:28.431 1 L 12 00010000-
9:16:28.431 1 L 12 00010000-SIPX end
```

•••

#### <Below is a RX SIP message>

```
9:16:28.431 1 L 12 00010000-SIPR begin (331 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:28.431 1 L 12 00010000-
                                 >SIP/2.0 100 Trying
 9:16:28.431 1 L 12 00010000-
                                 >FROM: "Dialogic Diva
SIPcontrol "<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.431 1 L 12 00010000-
                                 >TO: "Default"<sip:+17166395101@bufocs.local;user=phone>
 9:16:28.431 1 L 12 00010000-
                                 >CSEQ: 1 INVITE
 9:16:28.431 1 L 12 00010000-
                                 >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:28.431 1 L 12 00010000-
                                 >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.431 1 L 12 00010000-
                                 >CONTENT-LENGTH: 0
 9:16:28.431 1 L 12 00010000-
 9:16:28.431 1 L 12 00010000-SIPR end
9:16:28.665 0 L 12 00010000-SIPR begin (408 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:28.665 0 L 12 00010000-
                                 >SIP/2.0 183 Session Progress
 9:16:28.665 0 L 12 00010000-
                                 >FROM: "Dialogic Diva
SIPcontrol "<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.665 0 L 12 00010000-
                                 >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
 9:16:28.665 0 L 12 00010000-
                                 >CSEO: 1 INVITE
 9:16:28.665 0 L 12 00010000-
                                 >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:28.665 0 L 12 00010000-
                                 >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.665 0 L 12 00010000-
                                 >CONTENT-LENGTH: 0
 9:16:28.665 0 L 12 00010000-
                                 >SERVER: RTCC/3.0.0.0 MediationServer
 9:16:28.665 0 L 12 00010000-
                                 ~
 9:16:28.665 0 L 12 00010000-SIPR end
9:16:28.869 1 L 12 00010000-SIPR begin (399 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:28.869 1 L 12 00010000-
                                 >SIP/2.0 180 Ringing
 9:16:28.869 1 L 12 00010000-
                                 >FROM: "Dialogic Diva
SIPcontrol "<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.869 1 L 12 00010000-
                                 >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
 9:16:28.869 1 L 12 00010000-
                                 >CSEQ: 1 INVITE
                                 >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:28.869 1 L 12 00010000-
 9:16:28.869 1 L 12 00010000-
                                 >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.869 1 L 12 00010000-
                                 >CONTENT-LENGTH: 0
 9:16:28.869 1 L 12 00010000-
                                 >SERVER: RTCC/3.0.0.0 MediationServer
 9:16:28.869 1 L 12 00010000-
                                 >
 9:16:28.869 1 L 12 00010000-SIPR end
9:16:30.197 1 L 12 00010000-SIPR begin (836 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:30.197 1 L 12 00010000-
                                 >SIP/2.0 200 OK
                                 >FROM: "Dialogic Diva
 9:16:30.197 1 L 12 00010000-
SIPcontrol "<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:30.197 1 L 12 00010000-
                                 >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
 9:16:30.197 1 L 12 00010000-
                                 >CSEO: 1 INVITE
 9:16:30.197 1 L 12 00010000-
                                 >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:30.197 1 L 12 00010000-
                                 >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:30.197 1 L 12 00010000-
                                 >CONTACT:
<sip:dmg4000.BufOCS.local:5060;transport=Tcp;maddr=192.168.0.106>
 9:16:30.197 1 L 12 00010000-
                                 >CONTENT-LENGTH: 253
 9:16:30.197 1 L 12 00010000-
                                 >SUPPORTED: 100rel
 9:16:30.197 1 L 12 00010000-
                                 >CONTENT-TYPE: application/sdp; charset=utf-8
```

9:16:30.197	1	L	12	00010000-	>ALLOW: UPDATE
9:16:30.197	1	L	12	00010000-	>SERVER: RTCC/3.0.0.0 MediationServer
9:16:30.197	1	L	12	00010000-	>ALLOW: Ack, Cancel, Bye,Invite
9:16:30.197	1	$\mathbf{L}$	12	00010000-	>
9:16:30.197	1	L	12	00010000-	>v=0
9:16:30.197	1	L	12	00010000-	>o=- 0 0 IN IP4 192.168.0.106
9:16:30.197	1	L	12	00010000-	>s=session
9:16:30.197	1	L	12	00010000-	>c=IN IP4 192.168.0.106
9:16:30.197	1	L	12	00010000-	>b=CT:1000
9:16:30.197	1	L	12	00010000-	>t=0 0
9:16:30.197	1	L	12	00010000-	>m=audio 62438 RTP/AVP 8 101
9:16:30.197	1	L	12	00010000-	>c=IN IP4 192.168.0.106
9:16:30.197	1	L	12	00010000-	>a=rtcp:62439
9:16:30.197	1	L	12	00010000-	>a=label:Audio
9:16:30.197	1	L	12	00010000-	>a=rtpmap:8 PCMA/8000
9:16:30.197	1	L	12	00010000-	>a=rtpmap:101 telephone-event/8000
9:16:30.197	1	L	12	00010000-	>a=fmtp:101 0-16
9:16:30.197	1	L	12	00010000-	>a=ptime:20
9:16:30.197	1	L	12	00010000-SIPH	R end

<Bellow is a TX Q.931 ISDN message, after SIP session is established>

9:16:30.212 C 3 21:2391:136 - SIG-X(005) 08 02 80 17 07 Q.931 CR8017 CONN

Dialogic<sup>®</sup> 4000 Media Gateway Series Integration Note