

Dialogic[®] 4000 Media Gateway Series Integration Note Avaya 58500

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1. Scope

This document is intended to detail a typical installation and configuration of the Dialogic[®] 4000 Media Gateway Series if used to interface between a PBX and the Microsoft[®] 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

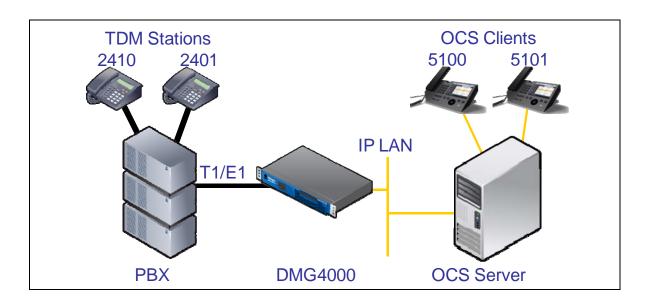
PBX Vendor	Avaya
Model(s)	S8500
Software Version(s)	Comm manager 3.0 – R013x.00.1.346.0
Additional Notes	N/A

2.2 Gateway

Gateway Model	Dialogic [®] 4000 Media Gateway Series
Software Version(s)	Dialogic [®] Diva [®] System Release software version 8.3.2 build 459 (formerly called Diva [®] Server software) Dialogic [®] Diva [®] SIPcontrol [™] 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[®] 4000 Media Gateway Series and OCS Server stands for Microsoft[®] 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, the TN464F DS1 interface card is needed.

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 most recent update to this document.

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Gateway Setup Notes 5.

Steps for setting up the gateway:

- Configuration of the Dialogic[®] Diva[®] Media Board drivers.
 Configuration of the Dialogic[®] Diva[®] SIPcontrol[™] software.

Dialogic[®] Diva[®] Media Board Configuration 5.1

The Diva Media Boards are configured in the Dialogic[®] Diva[®] Configuration Manager. To open the Configuration Manager, click:

Start > Programs > Dialogic Diva > Configuration Manager.

Note: In the Dialogic[®] Diva[®] 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 🛛 😵 😥			
✓ Status Bar	Property	Value	
Advanced	Line Type	Primary Rate Line (23 B-Channels)	
	Switch Type	Q-SIG T1)	J
Services diva	PBX Type	Generic	
API	Q-Sig Standard	Automatic	
0000	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.	1
	Configure the line-specific pr		
	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.			//

Note: The number of TDM circuits varies depending on the used Dialogic[®] 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[®] Media Gateway.

5.2 Dialogic[®] Diva[®] SIPcontrol[™] 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*[®] 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.

Back +	🗇 - 🗷 🔹 🏠 🔎 Sei	arch 😙 Favorites 🛛 🧔 🔹 😓							
iress	http://127.0.0.1:10005/sipcon	trol.cgi							
ය 👌 PB0	X IP Media Gateway								
Dial	ogic								
•		номе	S	Pcontrol Configuration					
(Configuration	PSTN Interface Configuration	ภา						۲
	SIPcontrol >	Name	Ne	Hardware Description	Channels	Dialplan	Er	nabled	
	Password	Controller1	1	Eicon Diva Server V-4PRI/E1/T1 - PORT 1 SN: 1	223 23	none	• 6	7	Detail
		Controller2	2	Eicon Diva Server V-4PRI/E1/T1 - PORT 2 SN: 1	223 23	none	• 6	7	Detail
Î	System	Controller3	3	Eicon Diva Server V-4PRI/E1/T1 - PORT 3 SN: 1	223 23	none	• 6	7	Detail
	Service Status	Controller4	4	Eicon Diva Server V-4PRI/E1/T1 - PORT 4 SN: 1	223 23	none	• 5	7	Detail
•	Licensing	Network Interface Configura	ition						۲
	License Management	Name	Devic	e	IP Address	Protocol	SIP List	en Port	Enable
		Intel(R) PR01000 EB Network Co	onn Intel(R) PRD/1000 EB Network Connection with I/D Acceleration	192.168.0.106	al 💌	(9803)		4
		Local Loopback Interface	Local I	Loopback Interface	127.0.0.1	al 💌	5060	1	Г
		RTP Start Port	30000						
		RTP End Port	39999	8					
		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[®] Mediation Server. Given that on these gateways the Microsoft[®] 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[®] 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[®] Mediation Server as shown below.

dit SIP Peer Configuration	
Name:	Mediation Server
Peer type:	MS 0CS 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:	
Alive check:	
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										(
Name	Default SIP to PSTN Peer	Host	Port	IP Protocol	Display Name to	Dialplan		Enabled		
Mediation Server	v	192.168.0.106	5060	TCP		none	٠	₹	Details	Delete
			Add	1						
and a family set of the set of the set										(
outing Configuration										
Name	Sources	Des	tinations				Addres	в Мар	Enabled	
Outbound Route	Mediation Server		troller1 (Si troller4 (Si		2 (Slave), Controller3	(Slave),	none	*	•	t L Details
Inbound Route	Controller1, Controller2, Con Controller4	ntroller3, Med	iation Sen	ver (Slave)			none	٠	•	t L Details

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[®] 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.

ttp://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
Address Normalization	
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) -
lone	internet

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[®] 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):	
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[®] 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[®] 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 [®] OCS	То РВХ
Internal	+1716639xxxx	716639xxxx

Called number	From Microsoft [®] 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				
Name	Rule Name	Stop on Match	Enabled	
	Calling number			↑ ↓ Details De
	Called - Internal		v	↑ ↓ Details De
Outbound	Called - National		v	↑ ↓ Details De
	Called - International	v	v	↑ ↓ Details De
	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:		
where <x> represents the number to "S" (subscriber) or empty (unknown)</x>	r those normalized via dialplan) are written as "≺X≻5551234", γpe 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:

Edit Address Map Configuration	
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:	
where <x> represents the number "S" (subscriber) or empty (unknown</x>	or 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

Edit Address Map Configuration		0
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:		
where <x> represents the number to "S" (subscriber) or empty (unknown)</x>	r those normalized via dialplan) are written as "«X≈5551234*, ype and may be either "+" (international), "N" (national), written as userinfo@domain.tid, 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 Address N	1ap - Microsoft Internet Explorer	_ 🗆 ×
		A
Edit Address Map Configuration		
Address map name:	Outbound	
Rule name:	Called - International	
Called address expression:	^\+	
Called address format:	9011	
Calling address expression:		
Calling address format:		
Redirect address expression:		
Redirect address format:		
Stop on match:	N	
where <x> represents the number t "S" (subscriber) or empty (unknown)</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	
	OK Cancel	
		7
One	📄 📄 📄 🔮 Internet	

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					6
Name	Sources	Destinations	Address Map	Enabled	
Outbound Route	Mediation Server	Controller1 (Slave), Controller2 (Slave), Controller3 (Slave), Controller4 (Slave)		v	f L Details
Inbound Route	Controller1, Controller2, Controller3, Controller4	Mediation Server (Slave)	none 💌	1	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 [®] OCS
Internal	xxxx (with subscriber type of number)	+1716639xxxx

Calling number	From PBX	To Microsoft [®] 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

Address Map Configura	ation			
Name	Rule Name	Stop on Match	Enabled	
	Calling number		$\overline{\mathbf{v}}$	↑ ↓ Details D
	Called - Internal	$\overline{\mathbf{v}}$		↑ ↓ Details D
Outbound	Called - National	$\overline{\mathbf{v}}$	V	↑ ↓ Details D
	Called - International	$\overline{\mathbf{v}}$		↑ ↓ Details D
	Add Rule			
	Called			↑ ↓ Details D
	Calling - Internal	$\overline{\mathbf{v}}$		↑ ↓ Details D
Inbound	Calling - National			î ↓ Details D
	Calling - International	V		↑ ↓ Details D
	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:	
where <x> represents the number "S" (subscriber) or empty (unknown</x>	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

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http://127.0.0.1:10005 - Edit Address	Map - Microsoft Internet Explorer	i i i i i i i i i i i i i i i i i i i	
Edit Address Map Configuration			\odot
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:			
"S" (subscriber) or empty (unknowr - Addresses received from SIP are headers.	written as userinfo@domain.tld, lik	e in the respective SIP	
	OK Cancel		
one		🔮 Internet	
ttp://127.0.0.1:10005 - Edit Address	Map - Microsoft Internet Explorer		
Edit Address Map Configuration			۲
Address map name:	Inbound		
Rule name:	Calling - National		
Called address expression:			

^N(\d{10})\$

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

🔹 🚺 🔮 Internet

+1\$1

Г Г

.

Calling address expression: Calling address format:

Redirect address expression:

Redirect address format:

Stop on match:

🙆 Done

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

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

Routing Configuration					¢
Name	Sources	Destinations	Address Map	Enabled	
Outbound Route	Mediation Server	Controller1 (Slave), Controller2 (Slave), Controller3 (Slave), Controller4 (Slave)	Outbound 💌	~	† L Details
Inbound Route	Controller1, Controller2, Controller3, Controller4	Mediation Server (Slave)	Inbound 💌	2	1 L Details
		Add			

5.2.5 Restarting the Dialogic[®] Diva[®] SIPcontrol[™] 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.

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6. PBX Setup Notes

The basic steps of setting up the PBX for use with this gateway and a voice messaging system are as follows:

- 1. Setting up system parameters and customer options.
- 2. Administrating T1 interface.
- 3. Programming AAR for trunk access.
- 4. Setting up method of call direction and coverage.

All PBX programming is done via a Windows[®]-based Java application. The basic commands that you will encounter on the PBX to perform these actions are:

change; delete; and add

6.1 Setting Up System Parameters and Customer Options

Use the display system-parameters customer options command such that several Q.SIG and trunk-specific parameters needed for the integration to operate properly are enabled.

Note that configuration parameters in this area that are not already enabled may require special maintenance or configuration by Avaya Customer support and/or may require additional software packages be installed to configure properly. Once the command has been entered, press the NEXT PAGE soft key until the Optional Features section is displayed as shown below.

	REFRESH F2	F3	F4	HELP F5	 F6	NEXT PA	PREV F8							
F1 lisplay sys	F2				F6									
display sys				F5	F6	F7								
	tem-parame	ters customer					10							
Abbrevi		display system-parameters customer-options Page 3 of 10 OPTIONAL FEATURES												
Abbrevi		OP	TIONAL	FEATURES										
	ated Diali	ng Enhanced L	iet? v	Audi	hle Message	Waiting?	v							
		ty Gateway (A	-		Authorizati		-							
				Backup Cluster			-							
	-	ling Start at	-	Laonap Labour		S Branch? :								
		Call Classif.	-			CAS Main? :	n							
ARS? y Change COR by FAC? n														
	ARS/2	AAR Partition	-	Computer Tele	-	-								
А	RS/AAR Dia	ling without :	FAC? n	-										
А	SAI Link Co	ore Capabilit.	ies? n	- (DCS	(Basic)?	y)							
А	SAI Link P	lus Capabilit.	ies? n		DCS Call	Coverage?	y							
Asyn	c. Transfe:	r Mode (ATM)	PNC? n		DCS with R	erouting?	У							
Async. Tr	ansfer Mode	e (ATM) Trunk.	ing? n		<u> </u>		~							
	ATM WAN	Spare Proces:	sor? n	Digital Los	s Plan Modi	fication?	У							
		A	TMS? y			DS1 MSP? :	n							
	Atte	endant Vector.	ing? y	DS	1 Echo Canc	ellation?	У							

Review the DCS (Basic), DCS Call Coverage, and DCS with Rerouting options. They must be set to y for this integration to work properly. If they are not enabled, then contact your Avaya support representative to have these features enabled.

Press the NEXT PAGE soft key, and the next page is displayed.

👙 Native Configur	ation Manager Ve	rsion 3.0-81 conn	ected to 10.242.2	02.91 : 5022						
File View										
Command: disp sys	cus						▼ Send			
CANCEL	REFRESH			HELP		NEXT PA	PREV PA			
F1	F2	F3	F4	F5	F6	F7	F8			
display sys	tem-paramet	ers custome	er-options		Pa	ige 4 of	10			
		o	PTIONAL FEF	TURES						
Emergency Access to Attendant? y IP Stations? y										
Enable 'dadmin' Login? y Internet Protocol (IP) PNC? y										
Enhanced Conferencing? y ISDN Feature Plus? n										
Enhanced EC500? y ISDN Network Call Redirection? n										
Enterpr	ise Surviva	able Server?	'n		ISDN-	BRI Trunks?	<u>' y</u>			
Ente	•	Licensing?				ISDN-PRI?	<u> </u>			
		nistration?		Local	. Survivable					
	-	J/Fwd Admin?	-			Call Trace?	-			
		larm Admin?	-		lia Encrypti		-			
Five Port		lax Per MCC?		e Code for (Centralized	Voice Mail?	У			
		le Billing?	-							
	-	count Codes?	-		tifrequency					
Globa			-		erver Interf					
	•	ty (Basic)?	-		Call Handli		-			
Hospitalit	y (G3V3 Enh	ancements)?	-	timedia Cal	l Handling	(Enhanced)?	У			
		IP Trunks?	У							
	TD Attondor	nt Consoles?								
			-	fort the pr	ermission ch					
(NO	in. iou mus	at rogori a	togin to ei	reet the pe	similarity of Ch	anges.)				
Info:										

Review the ISDN-PRI option. It must be set to y for this integration to work properly. If this option is not enabled, then contact your Avaya support representative to have these features enabled. Press the NEXT PAGE soft key, and the next page is displayed.

Stative Configur	ation Manager Ve	rsion 3.0-81 conn	ected to 10.242.2	02.91 : 5022						
Command: disp sys	cus						▼ Send			
CANCEL	REFRESH			HELP		NEXT PA	PREV PA			
F1	F2	F3	F4	F5	F6	F7	F8			
display sys	tem-paramet	ers custome	er-options		Pa	ge 5 of	10			
		C	OPTIONAL FEF	TURES						
Multiple I	evel Preced	ational Loc lence & Pree Multiple Loc	emption? n		Station and n as Virtual					
System Management Data Transfer? n										
E	Personal Station Access (PSA)? y Tenant Partitioning? y									
		Posted Me	essages? y	Termin	hal Trans. I	nit. (TTI)?	'У			
		PNC Dupli	cation? n		Time of D	ay Routing?	У			
	Por	t Network 🖇	Support? y		Uniform Di	aling Plan?	' Y			
				Usage Al	location En	hancements?	' y			
	Process	or and Syst	em MSP? n	TN2501	l VAL Maximu	m Capacity?	' y			
	(I	Private Netw	working? y							
	Ē	rocessor Et	hernet? y		Wideband	Switching?	' y			
						Wireless?	' y			
		Remote	Office? y							
R	estrict Cal	l Forward () ff Net? y							
	Seco	ndary Data	Module? y							
(ис	TE: You mus	t logoff &	login to ef	fect the pe	ermission ch	anges.)				
Info:										

Review the Private Networking and Uniform Dialing Plan options. They must be set to y for this integration to work properly. It they are not enabled, then you must contact you Avaya support representative to have these features enabled.

Press the NEXT PAGE soft key, until page #8 (QSIG Optional Features) is displayed.

Native Configur File View	ation Manager Ve	rsion 3.0-81 conn	ected to 10.242.20	2.91 : 5022								
Command: disp sys	cus						▼ Send					
CANCEL	REFRESH			HELP		NEXT PA	. PREV PA					
F1	F2	F3	F4	F5	F6	F7	F8					
display system-parameters customer-options Page 8 of 10 QSIG OPTIONAL FEATURES Basic Call Setup? y Basic Supplementary Services? y												
		Basi	.c Supplement	ary Servio	ces? y							
Centralized Attendant? y												
Interworking with DCS? y												
Supplementary Services with Rerouting? y												
Transfer into QSIG Voice Mail? y												
Value-Added (VALU)? y												
(NO	TE: You mus	t logoff &	login to eff	fect the pe	rmission	changes.)						
nfo:												

Review the Basic Call Setup, Basic Supplementary Services, Centralized Attendant, Interworking with DCS, Supplementary Services with Rerouting, Transfer into QSIG Voice Mail, and Value Added (VALU) options. They must be set to y for this integration to work properly. If they are not enabled, then you need to contact your Avaya support representative to have these features enabled.

Use the Change System-Parameter Features command to make some required modifications to the systems advanced ISDN and Q.SIG features. In order to make changes to the following fields, you must have administrative rights for the Avaya switch. If you do not, then you need to contact your Avaya support representative.

Press the NEXT PAGE soft key, until you get to the ISDN PARAMETERS page.

👙 Native Configu	ration Manager Ve	rsion 3.0-81 conne	cted to 10.242.3	202.91 : 5022						
File View										
Command: disp sys	fea						- Send			
CANCEL	REFRESH			HELP		NEXT PA	PREV PA			
F1	F2	F3	F4	F5	F6	F7	F8			
display sys	tem-paramet	ers feature	s		Pa	age 8 of	16			
		FEATURE-RI	ELATED SYS:	TEM PARAMETE	RS					
ISDN PARAME	TERS									
Send Non-ISDN Trunk Group Name as Connected Name? n										
Display Connected Name/Number for ISDN DCS Calls? n										
Send ISDN Trunk Group Name on Tandem Calls? y										
Send Custom Messages Through QSIG? y										
				· -						
		QSI	IG TSC Ext	ension: 2500						
MWI - Numb	er of Digit	s Per Voice	Mail Subso	criber: 4						
			ional CPN :							
		Internat	ional CPN :	Prefix:						
		Pass Pref:	ixed CPN to	o ASAI? y						
Unknowr	n Numbers Co	nsidered Int	ternal for	AUDIX? n						
	USNI Call	ing Name fo	r Outgoing	Calls? y						
	Path Re	placement w	ith Measur	ements? y	_					
		; Path Repla	cement Exte	ension: 2501	.]					
	Path Repla	ace While in	Queue/Vect	toring? y	_					
Info:										

- In the QSIG TSC Extension field enter XXXX
 - $\circ \quad$ where $\tt XXXX$ is any unused extension number.
- In the MWI-Number of Digits Per Voice Mail Subscriber field enter X o where X is the length of digits of the subscribers extensions.
- In the QSIG Path Replacement Extension field enter XXXX o where XXXX is any unused extension number.

Use the change system-parameters coverage-forwarding command to modify the system features relevant for forwarding the extensions and call coverage such that the Q.SIG supplementary services work properly.

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le View							
mmand: display	system-parameters	COV					💌 Seno
CANCEL	REFRESH			HELP		NEXT PA	PREV PA
F1	F2	F3	F4	F5	F6	F7	F8
	SYSTEM	ers coverag 4 PARAMETERS ING PARAMETE	CALL COVER	ng RAGE / CALL :		nge 1 of	2
Of	f-Net Cvg Sı	ibsequent Re Coverag	direction/G ge - Caller	CFWD No Ans CFWD No Ans Response In tion of Inco	Interval (r terval (sec	rings): 2 conds): 4	
		-	t for Trans) Held SBA a ferred Inco PROGRESS In Maintain	ming Trunk	Calls? n Mation? n	
	QSIG VALU	J Coverage O	-	IG Diversio Station Hunt		-	
FORWARDING					Forward Ove After Forwa		

• In the Maintain SBA At Principal field entery to enable this feature.

Before moving to the step, be sure to save your work.

6.2 Administrating the T1 Interface

Use the add ds1 command to setup the DS1 card with the proper settings. The command takes an argument that includes the cabinet and slot number where the DS1 card is located. For example:

Command = add ds1 all

Add a ds1 circuit card into the configuration that is located in cabinet 'a' and is in slot #11 of that cabinet.

👙 Native Configuration Manager Ve	ersion 3.0-81 co	nnected to 10.242.2	02.91 : 5022				
File View							
Command: disp DS1 a11							▼ Send
CANCEL REFRESH			HELP			NEXT PA	PREV PA
F1 F2	F3	F4	F5	F6		F7	F8
display ds1 a11					Pa	ge 1 of	2
		DS1 CIRCUIT	PACK				
Location:				lame: Ç	-		
Bit Rate:	1.544		Line Cod	ling: b	8zs		
Line Compensation:	1		Framing M	lode: e	sf		
Signaling Mode:	isdn-pri						
Connect:	pbx		Interf	ace: p	eer-m	aster	
TN-C7 Long Timers?	n		Peer Proto	col: Ç	2-SIG		
Interworking Message:	PROGress		8	ide: b))		
Interface Companding:	mulaw			CRC? n	ι		
Idle Code:	11111111						
	D	CP/Analog Bea	rer Capabil	ity: 3	.1kHz		
		I	303 Timer(s	ec): 4			
Slip Detection?	n	Near	-end CSU Ty	rpe: ot	her		
Info:							

- In the Name field, enter in any string to assign the DS1 a name
- In the Bit Rate field, enter 1.544. (The standard clock rate for T1)
- In the Line Coding field, enter b8zs.
- In the Line Compression field, enter 1.
- In the Framing Mode field, enter esf.
- In the Signaling Mode field, enter isdn-pri.
- In the Connect field, enter pbx.
- In the Interface field, enter peer-master.
- In the Peer Protocol field, enter Q-SIG.
- In the Interworking Message field, enter PROGress.
- In the Side field, enter b.
- In the Interface Companding field, enter mulaw.
- In the CRC field, enter n.
- In the Idle Code field, enter 11111111. (8 1's)

Press NEXT PAGE to move to the next page of settings.

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Se Native Configuration View	ation Manager Ver	rsion 3.0-81 conn	ected to 10.242.2	02.91 : 5022			<u> X</u>			
Command: disp DS1	a11						▼ Send			
CANCEL	REFRESH			HELP		NEXT PA	PREV PA			
F1	F2	F3	F4	F5	F6	F7	F8			
display ds1	a11				Pa	qe 2 of	2			
uispiuj usi	ull	DSI	L CIRCUIT PA	CK	14	.go 201	-			
ESF DATA LINK OPTIONS										
ESF DATA LINK OPTIONS Network Management Protocol: tabs Send ANSI-T1.403 One-Second Performance Reports? n Far-end CSU Address: b										
Info:										

• In the Far-end CSU Address field, enter b.

Before moving on to the next command, be sure to save your work.

Use the add signaling-group command to build a signaling group for the Q.SIG link to the gateway. This signaling group specifies the type of interface being used, the signaling channel (D-channel) wiring address for the T1 span being used, and the protocol used on the signaling channel.

	uration Manager Ve	rsion 3.0-81 conn	ected to 10.24	2.202.91 : 50	22			-	_ 🗆 🛛
File View									
Command:									 Send
CANCEL				HELF	C				
F1	F2	F3	F4	F5		F6	F7		F8
display si	gnaling-grou	ւթ 1							
		£	SIGNALING	GROUP					
Group Num	her 1	Gro	oup Type:	isdn-pri					
Oroup Num		sociated S:			M⊆	ix number of	NCA TSC.	10	
	11	Primary D-				lax number of			
		frimary D	ondinio1.	01111101		ik Group for			
Tru	nk Group for	Channel Se	election:	3				-	
	Supplementar								
		-							
Info:									

- In the Group Type field, enter isdn-pri.
- In the Associated Signaling field, enter y.
- In the Max number of NCA TSC field, enter 10.
- In the Primary D-Channel field, enter XXXX24
 - \circ where xxxx24 is the 24th port number of the T1 board.
- In the Max number of CA TSC field, enter 10.
- In the Trunk Group for Channel Selection field, enter X.
 - where x is the trunk group number created in the next step.
 Note: You will need to return to here and edit this number after you have created the trunk group.
- In the Supplementary Service Protocol field, enter b.

Use the add trunk-group command to add a new trunk group that will contain all the members of the ISDN line and tie them together with the signaling group created above.

🖢 Native Configuration Manager Version 3.0-81 connected to 10.242.202.91 : 5022											
File View											
Command: disp tru	3						▼ Send				
CANCEL	REFRESH			HELP		NEXT PA	PREV PA				
F1	F2	F3	F4	F5	F6	F7	F8				
display tru	ink-group 3				Pa	ige 1 of	20				
		г	RUNK GROUP								
Group Number: 3 Group Type: isdn CDR Reports: y											
Group Nam	le: qsig)R: 1	TN: 1	TAC: 83					
Directio	n: two-way	Outo	joing Displa	ay? y	Carrier M	ledium: PRI/	BRI				
Dial Access? y Busy Threshold: 99 Night Service:											
Queue Length: O											
Service Type: tie Auth Code? n TestCall ITC: rest											
		Far End	Test Line N	Io:							
TestCall BC	C: 4										
TRUNK PARAM	ETERS										
Co	deset to Se	end Display:	6 Code	eset to Send	d National I	Es: 6					
Мах	: Message Si	ze to Send:	260 Char	ge Advice:	none						
Supplemen	tary Servic	e Protocol:	b Digi	t Handling	(in/out): e	nbloc/enblo	c				
	Trunk Hunt	: ascend		QS	SIG Value-Ad	lded? n					
				Digi	ital Loss Gr	oup: 13					
Incoming Ca	lling Numbe	er - Delete:	Insert	:	Form	at: unk-unk	:				
	Bit Rate	: 1200	Synchro	nization: a	async Dup	lex: full					
Disconnect	Supervisio	on - In? y	Out? n								
Answer Sup	ervision Ti	.meout: O									
Info:											

- In the Group Type field, enter isdn.
- In the Group Name field, enter a text string to give the group a name.
- In the COR field, enter the number of a class of restriction to be used, if there is going to be one. Note: Setting any restrictions via a COR can effect how the trunk operates so these should be as liberal as possible.
- In the TN field, enter the proper tenant number.
- In the TAC field, enter XX
 - where xx is an available DAC (used as a trunk access code) number from the dial-plan.
- In the Direction field, enter two-way.
- In the Outgoing Display field, enter y.
- In the Carrier Medium field, enter PRI/BRI.
- In the Service Type field, enter tie.
- In the Supplementary Service Protocol field, enter b.
- In the Digit Handling (in/out) field, enter enbloc/enbloc.
- In the Format field, enter unk-unk.

Press NEXT PAGE to advance to the next page of settings.

👙 Native Configu	ration Manager Ve	rsion 3.0-81 conn	ected to 10.242.2	02.91 : 5022				. 🗆 🗙
File View								
Command: disp tru	3						-	Send
CANCEL	REFRESH			HELP		NEXT PA	PREV	PA
F1	F2	F3	F4	F5	F6	F7	F8	•
display tru	ink-group 3				Pa	ige 2 of	20	
TRUNK FEATU	IRES							
F	CA Assignme	ent? n	Measu	red: none	Wideban	d Support?	n	
			Internal Al	.ert? n	Maintena	ince Tests?	У	
		Da	ata Restrict	ion? n	NCA-TSC Tru	ink Member:	23	
			Send N	Jame: n	Send Calli	ng Number:	У	
	Used for I	DCS? n	Нор	Dgt? n				
Suppress	s # Outpulsi	ing?n Fo	ormat: unk-p	vt				
Outgoing C	Channel ID H	Incoding: p	referred	UUI IE Tre	eatment: ser	vice-provid	der	
				Replac	ce Restricte	d Numbers?	n	
				Replace	e Unavailabl	e Numbers?	n	
				\$	Send Connect	ed Number:	У	
				Hold/	'Unhold Noti	fications?	У	
	Send UUI	IE? y		Modify 1	Fandem Calli	ng Number?	n	
	Send UC	CID? n						
Send Codes	set 6/7 LAI	IE? y		Dsi	l Echo Cance	llation? n		
		Neta	work (Japan)	Needs Conr	nect Before	Disconnect'	? n	
Info:								
J								

- In the NCA-TSC Trunk Member field, enter 23.
- In the Send Calling Number field, enter y.
- In the Format field, enter unk-pvt.
- In the Outgoing Channel ID Encoding field, enter preferred.
- In the Send Connected Number field, enter y.
- In the Send UUI IE field, enter y.

Press NEXT PAGE to advance to the next page of settings.

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불 Native Configura	ition Manager Ver	sion 3.0-81 conne	cted to 10.242.2	02.91 : 5022			
File View							
Command: disp tru 3							▼ Send
CANCEL	REFRESH			HELP		NEXT PA	PREV PA
F1	F2	F3	F4	F5	F6	F7	F8
display tru	nk-group 3	QSIG TRUNK	GROUP OPTI	ONS	Pa	ge 3 of	20
I	Diversion b	y Reroute?	У				
		placement?					
Path Replace	ement with	Retention? :	n				
Patł	n Replaceme	nt Method: 3	better-rout	e			
		SBS? :	n				
Info:							

- In the Diversion by Reroute field, enter y.
- In the Path Replacement field, enter y. Note: This parameter enables the path replacement functionality of the Q.SIG supplemental services package.
- In the Path Replacement Method field, you can enter a selection to tune how your path replacement optimization is done.

Press NEXT PAGE to advance to the next page of settings.

On this page, enter each bearer channel of the T1 span, leaving off the D-channel since that is connected to the group via the Signaling Group and is not used to carry voice.

👙 Nativ	e Configurati	ion Manager	Version 3	.0-81 con	inected to 10.242.2	02.91 : 5022			
File Vie	W								
Comman	d: disp tru 3								▼ Send
CA	NCEL	11 TN464 F 12 TN464 F 13 TN464 F 14 TN464 F 15 TN464 F 16 TN464 F 17 TN464 F 18 TN464 F 19 TN464 F 10 TN464 F			HELP		NEXT PA	PREV PA	
	F1	F2		F3	F4	F5	F6	F7	F8
displ	Lay trun	c-group	3				Pa	ıqe 4.of	20
-	-				TRUNK GROUP			-	
					Admini	stered Memb	ers (min/ma	ax): 1/2	3
GROUE	MEMBER	ASSIGN	4ENTS		To	tal Adminis	tered Membe	rs: 23	
			_						
	Port			me	Night	-	Grp		
1:	01A1101					1			
2:	01A1102		-			1			
3:	01A1103		-			1			
4:	01A1104		-			1			
5:	01A1105		-			1			
6:	01A1106		F			1			
7:	01A1107		F			1			
8:	01A1108		F			1			
9:	01A1109	TN464	F			1			
10:	01A1110	TN464	F			1			
11:		TN464	F			1			
12:	01A1112	TN464	F			1			
	01A1113	TN464	F			1			
14:	01A1114	TN464	F			1			
15:	01A1115	TN464	F			1			
Info:									

- In each of the Port fields, enter the port number that designates the physical wiring address of each of the bearer channels in the span.
- In each of the corresponding Sig Grp fields, for each port enter in the number of the Signaling Group created above.

6.3 **Programming AAR for Trunk Access**

Important Note: In order for Path Replacement to work on Join transfers, the inbound calls MUST use an AAR number to access the trunk, and NOT the assigned Trunk Access Code (TAC). If an outgoing call is made using a TAC, or a call was extended by an attendant using DTGS (Direct Trunk Group Selection), the user has intentionally chosen a particular Trunk Group for the outgoing call, and the PBX will not use Path Replacement optimization in these cases.

Use the change uniform-dialplan command to setup an extension that can be used to access the trunks via AAR.

👙 Native Configur	ation	Manag	er Versio	n 3.0-81	connect	ed to 10).242.2	02.91 : 5	022						
File View															
Command: change	uniforn	n-dialpl	an 5												 Send
CANCEL	R	EFRESH	1	ENTER		CLEAR	Fl	HEL	.P			NE	XT PA	Р	REV PA
F1		F2		F3		F4		F5	i		F6		F7		F8
change unif	orm-	-dia	lplan	5							Pa	age	1 of	2	
			υ	NIFORM	1 DIA	L PLA	N TAE	LE							
											Per	cent	Full:	0	
Matching			Inser	t		Node	Ma	tching	r		Insert			Node	
Pattern	Len	Del	Digit	s Net	Conv	Num	Рa	ttern	Len	Del	Digits	Net	Conv	Num	
5	4	0		aar	n								n		
					n								n		
					n								n		
					n								n		
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					n								n		
					n								n		
					n								n		
					n								n		
Info: Right-click in a	a field to) see va	alid entries	s or help te	xt.										

- In the Matching Pattern field, enter a number to define as a dialable AAR number. In this example it is 5.
- In the Len field, enter X
 - \circ where x is the length of your dialable number. In this example 4.
- In the Del field, enter 0.
- In the Net field, enter aar.
- In the Conv field, enter n.

Use the change route-pattern command to define a route rule pattern that when referenced will direct calls to the trunk group you created above.

🍨 Native Configur	ation Manager	Version 3.0-81 c	onnected to 10.2	42.202.91 : 5022			
File View							
Command: cha rou ′	15						 Send
CANCEL	REFRESH	ENTER	CLEAR FI	HELP		NEXT PA	PREV PA
F1	F2	F3	F4	F5	F6	F7	F8
change rout	e-patterr	n 15			Pa	age 1 of	3
			mber: 15 E CCAN? n	attern Name: Secure SIP?			
GrpFRL	NPA Pfx	Hop Toll No	o. Inserte	d		DCS/	IXC
No	Mrk	Lmt List De	el Digits			QSIG	
		Dg	gts			Intw	
1: 3 0						У	
2:						n	
3:						n	
4:						n	
5:						n	
6:						n	
BCC VA	LUE TSC	CA-TSC 3	ITC BCIE Se	rvice/Feature	e PARM No. I	Numbering I	JAR
0123	4 W	Request			Dgts 1	Format	
					Subaddres	ss	
1: уууу	yn y	as-needed	rest			unk-unk i	none
2: уууу	yn n		rest			1	none
3: уууу	yn n		rest			1	none
4: y y y y	yn n		rest			1	none
5: уууу	yn n		rest			1	none
6: уууу	yn n		rest			1	none
Info: Right-click in a	field to see vali	d entries or help text					

- In the Grp field, enter X.
 - where x is the number of the trunk group assigned above.
- In the FRL field, enter 0.
- In the Intw field, enter y.
- In the CA-TSC Request field, enter as-needed.
- In the Numbering Format field, enter unk-unk.

Use the change AAR Digit Analysis Table command to set up AAR to bind your dialable number setup in UDP to the route pattern you have just defined.

mand: change	e aar analysis 5							 Send
CANCEL	REFRESH	ENTER	CLEAR FI	HELP			NEXT PA	PREV PA
F1	F2	F3	F4	F5		F6	F7	F8
ange aar	analysis 5					Pa	age 1 of	2
		AAR	DIGIT ANALY	SIS TABI	ЪЕ			
						Percei	nt Full:	0
	Dialed	Total	Route	Call	Node	ANI		
	String		ax Pattern	Type	Num	Reqd		
5	atring		.2 15	aar	Num	n		
						n		
		-	_			n		
		-				n		
		-				n		
						n		
						n		
						n		
						n		
						n		
						n		
						n		
						n		
			_			n		
						n		

- In the Dialed String field, enter the number you defined in UDP as the AAR number. In this example we use 5.
- In the Min and Max fields, enter the length of the number you are going to dial. In this example 4 for the Min and 12 for the Max.
- In the Route Pattern field, enter the number of the route pattern that you defined above. In this example 2.
- In the Call Type field, enter aar.
- In the ANI Reqd field, enter n.

Use the busyout board and the release board commands to reset the trunk device that has been defined. When a T1 board is initially programmed, by default it is in a busy state. In order to enable the T1 board, it must be set to busy and then released.

mand: bus bo	a11					🔻 Se
CANCEL					NEXT PA	
F1	F2	F3 F4	F5	F6	F7	F8
syout bo	ard all				Page	1
		COMMAND RESUL	LTS			
Port	Maintenance N	ame Alt. Name	Result	Erro	r Code	
01A11	UDS1-BD		PASS			
01A1101	ISDN-TRK	0003/001	PASS			
01A1102	ISDN-TRK	0003/002	PASS			
01A1103	ISDN-TRK	0003/003	PASS			
01A1104	ISDN-TRK	0003/004	PASS			
01A1105	ISDN-TRK	0003/005	PASS			
01A1106	ISDN-TRK	0003/006	PASS			
01A1107	ISDN-TRK	0003/007	PASS			
01A1108	ISDN-TRK	0003/008	PASS			
01A1109	ISDN-TRK	0003/009	PASS			
01A1110	ISDN-TRK	0003/010	PASS			
01A1111	ISDN-TRK	0003/011	PASS			
01A1112	ISDN-TRK	0003/012	PASS			
01A1113	ISDN-TRK	0003/013	PASS			
01A1114	ISDN-TRK	0003/014	PASS			

Avaya S8500

mand: <mark>ststa</mark> tr	unk 3						- Se
CANCEL						NEXT PA	
F1	F2	F3	F4	F5	F6	F7	F8
lease bo	ard all					Page	1
		COM	MAND RESUI	JT S			
Port	Maintenance	Name Al	t. Name	Result	Error	Code	
01A11	UDS1-BD			PASS			
01A1101	ISDN-TRK	00	03/001	PASS			
01A1102	ISDN-TRK	00	03/002	PASS			
01A1103	ISDN-TRK	00	03/003	PASS			
01A1104	ISDN-TRK	00	03/004	PASS			
01A1105	ISDN-TRK	00	03/005	PASS			
01A1106	ISDN-TRK	0.0	03/006	PASS			
01A1107	ISDN-TRK	0.0	03/007	PASS			
01A1108	ISDN-TRK	0.0	03/008	PASS			
01A1109	ISDN-TRK	0.0	03/009	PASS			
01A1110	ISDN-TRK	0.0	03/010	PASS			
01A1111	ISDN-TRK	0.0	03/011	PASS			
01A1112	ISDN-TRK	0.0	03/012	PASS			
01A1113	ISDN-TRK	0.0	03/013	PASS			
01A1114	ISDN-TRK	0.0	03/014	PASS			

In both commands, you can use NEXT PAGE to view the trunk channels that are not visible on the page.

7. Microsoft[®] Office Communications Server 2007 (OCS) Setup

7.1 Steps for configuring Microsoft[®] OCS

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

If a Microsoft[®] 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[®] 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	1			_
<u>D</u> escription:		,				
test					-	-
Normalizatio	n Rules —					
		cessed in list (order: pla	asa usa tha l	buttons on the	
side for adju	on rules are pro- usting the order.	cessed in list (order; pla	ease use the l	Duttons on the	
- Europ					-	
5xxx 3xxx					-	
Зххх 2ххх						
Зххх					Up	
Зххх 2ххх					<u>⊔</u> p D <u>o</u> wn	
Зххх 2ххх						
Зххх 2ххх						
Зххх 2ххх				Denser		
Зххх 2ххх	<u>A</u> dd	<u>E</u> dit.		<u>R</u> emove		

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

dit Phone Number	Normaliz	ation	Rule				
<u>N</u> ame:		4 digits	5				
Click to copy an exis	ting rule.						<u>С</u> ору
Description:							
any 4 digits							
							•
Translation							
Phone pattern reg	ular expres:	sion:					
^(\d{4})\$							
<u>T</u> ranslation pattern	regular ex	pressio	on:				
+1716639\$1							
Click Helper for as regular expression:				n ph	one number		<u>H</u> elper
Test translation							
To test the translat pattern, the transla				umb	er. If it matc	hes	the phone
Sample dialed num	iber:						
, T <u>r</u> anslated number	:						
,				_			
			0K		Cancel		Help

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[®] Mediation Server. If you need to route some calls to a different Microsoft[®] Mediation Server, configure the Target phone numbers field accordingly.

To configure Microsoft[®] 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
	get phone number regu r more phone usages.	llar expressior), one or more
gateways, and one o	r more priorie usages.		
Target phone numb			
<u>T</u> arget regular expr	ession		
^\+?(\d*)\$			
			Helper
Address			
dmg4000.Buf0CS	6.local:5061		
1		<u>лл</u> 1	Demons
		<u>A</u> dd	<u>R</u> emove
Phone usages			
Default Usage			
1			
			<u>C</u> onfigure
	ОК		

This entry routes numbers with or without "+" prefix followed by any digits to Microsoft[®] 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[®] OCS client (such as Microsoft[®] Office Communicator) is in E.164 format, Microsoft[®] OCS must find a normalization rule to convert the dialed number to E.164. If no rule is found

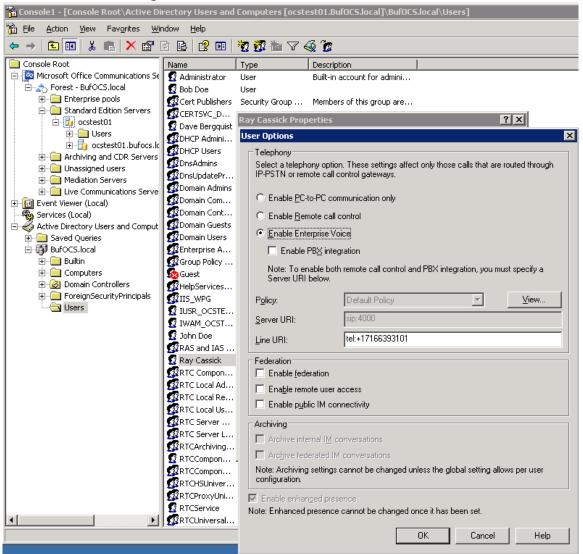
or matched, outbound calls will fail. In this case, Dialogic[®] Diva[®] Diagnostics trace will not receive an outbound SIP message, since the call will not yet have reached the SIP gateway.

7.2 Steps for configuring Microsoft[®] Office Communications Server 2007 (OCS) clients

🚡 Console1 - [Console Root\Active Dir	rectory Users and	Computers [ocste	st01.BufOC5.local]\BufOC9	5.local\Users]
🚡 Eile Action View Favorites Win	idow <u>H</u> elp			
	2 🗈 😫 🖬	🦉 📆 🖆 💎 4	ã 🐌	
Console Root				1
Console Root É Microsoft Office Communications Se	Name Administrator	Type User	Description Built-in account for admini	
Forest - BufOCS.local	S Bob Doe	User	Duilt-In account for authini	
Enterprise pools	Cert Publishers	Security Group	Members of this group are	
E Standard Edition Servers	CERTSVC_D		2,	
🖻 🚺 ocstest01	Dave Bergguist	Ray Cassick Prop	erties	? ×
🕀 🛄 Users		Member Of	Dial-in Enviro	nment Sessions I
in the second s	DHCP Users	· · · · · ·	- 1 I I I I I I I I I I I I I I I I I I	1
Archiving and CDR Servers		l }	Terminal Services Profile	Telephones Organization COM+ Communications
⊡ ⊡ Unassigned users	DnsUpdatePr	Remote control	· ·	00.111
	Domain Admins	Enable use	er for Office Communications Se	rver
Event Viewer (Local)	Domain Com	Sign-in name:		
Services (Local)	🕵 Domain Cont	sip:rcassic	k @ BufOC	
Active Directory Users and Comput	🕵 Domain Guests	Isip.icassi	ok @ BufOC	5.local
🗄 📄 Saved Queries	🕵 Domain Users	Server or pool:		
🖻 🖓 BufOCS.local	🕵 Enterprise A	ocstest01	.BufOCS.local	▼
🔁 🖓 🧰 Builtin	🕵 Group Policy	- Meetings-		
🗄 💼 Computers	5 Guest		onymous participants	
	MelpServices			
ForeignSecurityPrincipals	🕵 IIS_WPG	Policy:	Default Policy	<u> </u>
L	1USR_OCSTE			View
	IWAM_OCST			
	John Doe		ng settings cannot be changed	unless the global setting
	RAS and IAS	allows per u	ser configuration.	
	🖸 Ray Cassick			
	RTC Compon	A 1 192 1 1		Carlana
	RTC Local Ad	Additional optic	ons:	Configure
	RTC Local Re	<u> </u>		
	RTC Local Us RTC Server			
	RTC Server			
	RTC Server L		OK Cancel	Apply Help
	RTCCompon			
	RTCCompon	Security Group	Members can be used as	
	RTCHSUniver		Members can be used as	
	RTCProxyUni		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[®] 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[®] Diva[®] SIPcontrol[™] 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[®] OCS will ring the user Ray Cassick if he is logged on to Microsoft[®] OCS from Microsoft[®] Office Communicator or any Microsoft[®] OCS supported device.

8. Microsoft[®] Mediation Server Installation and Configuration

8.1 Installation

The gateways of the Dialogic[®] 4000 Media Gateway Series (DMG4000 Gateways) are shipped with pre-installed Microsoft[®] Mediation Server software. You can complete the Microsoft[®] Mediation Server configuration by running Microsoft[®] Office Communications Server 2007 (OCS) "Setup.exe" in the DMG4000 Gateways. In the Microsoft[®] 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[®] Mediation Server software.

Step 2: Activate Microsoft[®] 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[®] 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[®] 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[®] 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.

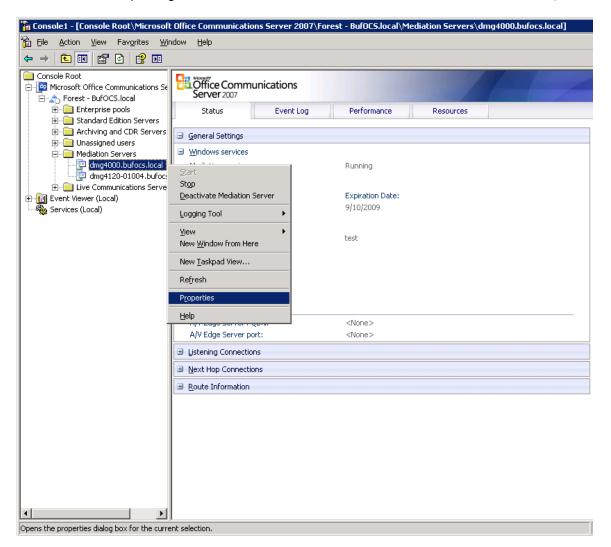
Eile Action View Favorites Window E	telp			
→ 🗈 🖪 🗼 🖻 🗙 📽 😫	Þ			
Console Root	Issued To 🔺	Issued By	Expiration Date	Intended Purposes
💼 Event Viewer (Local)	Equifax Secure Certificate Authority	Equifax Secure Certificate Authority	8/22/2018	Secure Email, Server
🍓 Services (Local)	🕮 Equifax Secure eBusiness CA-1	Equifax Secure eBusiness CA-1	6/20/2020	Secure Email, Server
🔘 Diva Server Management	🔛 Equifax Secure eBusiness CA-2	Equifax Secure eBusiness CA-2	6/23/2019	Secure Email, Server
🗄 📲 Status	🖼 Equifax Secure Global eBusiness C	Equifax Secure Global eBusiness CA-1	6/20/2020	Secure Email, Server
Active Connections	EUnet International Root CA	EUnet International Root CA	10/2/2018	Secure Email, Server
Call History	🕮 FESTE, Public Notary Certs	FESTE, Public Notary Certs	1/1/2020	Secure Email, Server
Active Directory Users and Computers [ocst	🖼 FESTE, Verified Certs	FESTE, Verified Certs	1/1/2020	Secure Email, Server
Certificates (Local Computer)	🖼 First Data Digital Certificates Inc	First Data Digital Certificates Inc. Ce	7/3/2019	Server Authenticatio
⊕ Personal F Trusted Root Certification Authorities	FNMT Clase 2 CA	FNMT Clase 2 CA	3/18/2019	Secure Email, Server
Certification Authorities	🖼 GlobalSign Root CA	GlobalSign Root CA	1/28/2014	Secure Email, Server
	GTE CyberTrust Global Root	GTE CyberTrust Global Root	8/13/2018	Secure Email, Client
Contempose must The prise must The prise must The prise must The prise must	GTE CyberTrust Root	GTE CyberTrust Root	4/3/2004	Secure Email, Client
	GTE CyberTrust Root	GTE CyberTrust Root	2/23/2006	Secure Email, Client
	Http://www.valicert.com/	http://www.valicert.com/	6/25/2019	Secure Email, Server
Third-Party Root Certification Authoritie:	Part Anter A	http://www.valicert.com/	6/25/2019	Secure Email, Server
Trusted People	Balttp://www.valicert.com/	http://www.valicert.com/	6/25/2019	Secure Email, Server
Enrollment Requests	IPS SERVIDORES	IPS SERVIDORES	12/29/2009	Secure Email, Server
E SPC	Microsoft Authenticode(tm) Root	Microsoft Authenticode(tm) Root Au	12/31/1999	Secure Email, Code S.
Microsoft Office Communications Server 200	Microsoft Root Authority	Microsoft Root Authority	12/31/2020	<all></all>
	Microsoft Root Certificate Authority	Microsoft Root Certificate Authority	5/9/2021	<all></all>
	NetLock Expressz (Class C) Tanusi	NetLock Expressz (Class C) Tanusity	2/20/2019	Server Authenticatio
	NetLock Kozjegyzoi (Class A) Tan	NetLock Kozjegyzoi (Class C) Tanusit	2/19/2019	Server Authenticatio
	NetLock Uzleti (Class B) Tanusitva	NetLock Uzleti (Class B) Tanusitvany	2/20/2019	Server Authenticatio
	NO LIABILITY ACCEPTED, (c)97 V	NO LIABILITY ACCEPTED, (c)97 Veri	1/7/2004	Time Stamping
	CCSTest	OCSTest	9/7/2012	<all></all>
	PTT Post Root CA	PTT Post Root CA	6/26/2012	Secure Email, Server
	Saunalahden Serveri CA	Saunalahden Serveri CA		,
			6/25/2019	Secure Email, Server
	Saunalahden Serveri CA	Saunalahden Serveri CA	6/25/2019	Secure Email, Server
	Secure Server Certification Autho	Secure Server Certification Authority	1/7/2010	Server Authentication
	SecureNet CA Class A	SecureNet CA Class A	10/16/2009	Secure Email, Server
	SecureNet CA Class B	SecureNet CA Class B	10/16/2009	Secure Email, Server
	SecureNet CA Root	SecureNet CA Root	10/16/2010	Secure Email, Server
	SecureNet CA SGC Root	SecureNet CA SGC Root	10/16/2009	Secure Email, Server
	SecureSign RootCA1	SecureSign RootCA1	9/15/2020	Secure Email, Server
p	SecureSign RootCA2	SecureSign RootCA2	9/15/2020	Secure Email, Server

- 4. Create the certificate request for the Microsoft[®] 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[®] Mediation Server is correct, then run the Deployment Wizard again or click Certificates in Available tasks in Microsoft[®] Mediation Server MMC snap-in.

8.2 Configuration

From the MMC snap-in, right-click the detected Microsoft[®] Mediation Server and select Properties.



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

dmg4000.bufocs.local Properties	X							
General Next Hop Connections Certificate	1							
Mediation Server								
EQDN: dmg4000.BufOCS.local								
⊆ommunications Server listening IP address:								
192.168.0.106								
Gateway 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	s 🗙
General Next Hop Connections	Certificate
Office Communications Server n Specify the Office Communication PSTN calls.	next hop ons Server used for routing inbound
EQDN:	
ocstest01.BufOCS.local	
Port:	5061
PSTN Gateway next hop	
Specify the PSTN gateway conn	nected to this server.
IP address:	192 . 168 . 0 . 106
Port:	9803
ОКС	Cancel <u>Apply</u> Help

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

Click the Certificate tab.

dmg4000.bufocs.local Properties	×							
General Next Hop Connections Certificate								
Server Certificate Specify the certificate to be used for inbound and outbound connections on this server.								
Issued to: dmg4000.BufOCS.local Issued by: OCSTest Valid from 9/11/2007 3:54 PM to 9/10/2009 3:54 PM.								
Select <u>C</u> ertificate								
Warning: Changing the certificate will have no effect on existing connections. Existing connections will continue to use the old certificate.								
OK Cancel Apply Help								

Select the certificate that will be used to communicate with Microsoft[®] OCS. Microsoft[®] 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[®] 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 [®] OCS client.	
2	Direct call from Microsoft [®] 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[®] Diva[®] Diagnostics tool: Used to review and analyze all SIP and ISDN traffic that relates to calls going into and leaving the Dialogic[®] 4000 Media Gateway.

10.2 Using the Dialogic[®] Diva[®] Diagnostics Tool

Before using the Dialogic[®] Diva[®] Diagnostics tool, you would need to enable it by setting the Dialogic[®] Diva[®] SIPcontrol[™] 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.

jle <u>E</u> dit ⊻iew <u>T</u> racing <u>H</u> elp			
🖬 👺 🔛 🤋 📢 🔤 🗅 🖪 🗉	🗹 🗙 🔹 🛎 🕼	D	Trace inactive, 132 Kbytes available.
Diva Tracing	Category	ID	Description
🗄 🛄 Trace Masks	D-Channel	D	Displays D-channel data as raw hex frames.
Drivers	🗹 Layer 1	1	Displays layer 1 state changes.
Maintenance Driver	Call Comments (SIG)	С	Displays decoded information for signaling frames and states
Combined Adapter Driver	Low Layer	0	Displays decoded low layer information
RAS (WAN Miniport)	Network Layer	N	Displays B-channel establishment and data indication.
CAPI Driver Driva Server Adapters	Data Link Error	F	Displays layer 2 link errors.
4PRI/E1/T1 #1223 Line 1	Miscellaneous	S	Textual information on call states etc.
	Extended	s	Displays module-specific detailed information.
	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[®] Diva[®] 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[®] Diva[®] Diagnostics traces for an inbound (TDM to IP) call to Microsoft[®] 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:+171663924010192.168.0.106:9803>
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
                                 >a=rtpmap:8 PCMA/8000
9:16:28.431 1 L 12 00010000-
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	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[®] 4000 Media Gateway Series Integration Note