

1. Scope

This document is intended to detail a typical installation and configuration of Dialogic[®] 2000 Media Gateway Series (DMG2000) when used to interface between PBX and Microsoft® Office Communications Server 2007 (OCS) application.

2. **Configuration Details**

Listed below are the specific details of the PBX and gateways used in the testing to construct the following documentation.

2.1 **PBX**

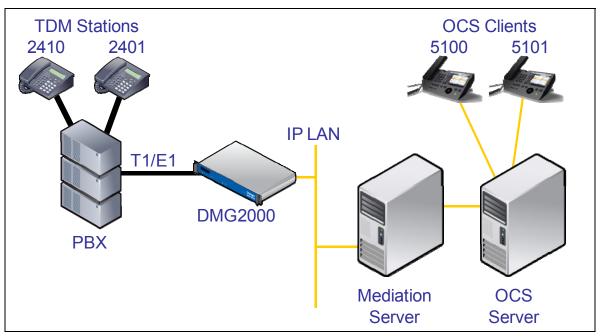
PBX Vendor	Siemens Hipath
Model	4000
Software Version	Version 2.0 SMR9 SMP0
Additional Notes	N/A

2.2 **Gateway**

Gateway Model	Dialogic® 2000 Media Gateway Series (DMG2000)
Software Version	6.0 (6.0.103)
Protocol	T1 QSIG

2.3 **System Diagram**

The diagram below details the setup used in the testing and creation of the technical document.



3. Prerequisites

3.1 PBX Prerequisites

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

3.1.1 PBX Equipment Required

To support the T1 QSIG configuration as documented you need the ISDN T1- DIU2U Q2216 interface card.

3.1.2 PBX Cabling Requirements

Cabling for QSIG connections must be CAT5e or better. Standard voice quality cable will not provide optimum signal quality and the gateway will have problems establishing connection on the D-Channel.

3.2 Gateway Prerequisites

The gateway needs to support a T1 QSIG interface.

4. Summary of Limitations

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

5. Gateway Setup Notes

Steps for setting up the gateway:

- Parameter Configuration
- Routing Engine Configuration

5.1 Parameter Configuration

To get the gateway connected between the PBX and mediation server there are only a few configuration options that are required.

During the initial setup of the Dialogic gateway using the serial port you must:

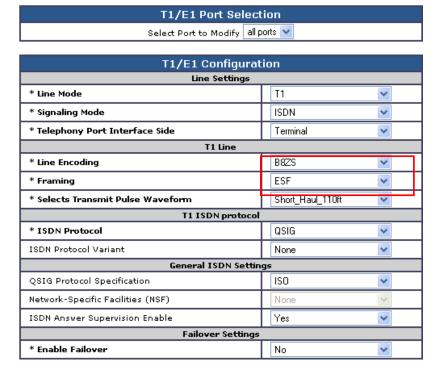
- Assign LAN 1 on the gateway a unique IP address, subnet mask and network gateway address (if the latter is required).
- Configure the gateway to use the SIP VoIP protocol.
- Set the Line Mode to T1.
- Set the Protocol to ISDN QSIG.

During the solution specific setup of the Dialogic gateway using the web interface you must:

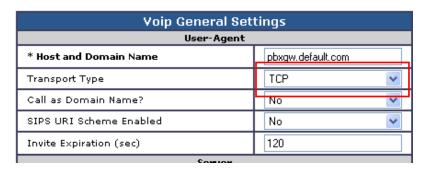
- In the IP Settings page:
 - o Set the BOOTP Enabled parameter to 'No'. (the default is Yes)

IP Settings, LAN1				
MAC	00-0e-0c-ab-d2-3e			
* Client IP Address	192.168.1.2			
* Client Subnet Mask	255.255.255.0			
* Default Network Gateway Address	192.168.1.250			
* BOOTP Enabled	No 💌			
* SNTP Server IP Address				
<u> </u>				

- ID Cottings | AM
- In the T1/E1 General page:
 - Set the Line Encoding and Line Framing as required by your T1 Interface.
 Typical settings are Encoding = B8ZS and Framing = ESF.



- In the VoIP General page:
 - Set the Transport Type parameter to TCP (the default is UDP)



- In the VoIP Media page:
 - o Set the RTP Fax/Modem Tone Relay Mode parameter to 'In band-Tone' (the default is RFC2833)
 - o Set the Signaling Digit Relay Mode parameter to 'Off' (the default is On)
 - o Set the Voice Activity Detection parameter to 'Off' (the default is On)

VoIP Media Settings						
	Audio					
* Audio Comp	pression	G.711u/G.711a				
RTP Digit Rel	ay Mode	RFC2833				
RTP Fax/Mod	em Tone Relay Mode	Inband-Tone 🔻				
* RTP Source	IP Address Validation	Off V				
* RTP Source	UDP Port Validation	Off				
Signaling Digi	t Relay Mode	Off				
Voice Activity	Detection	Off				
RFC 3960 Ear	ly Media Support	OnDemand 💌				
Codec	Frame Size	Frames per Packet				
G.711	30	1				
G.723.1	30	1				
G.729AB	10	3				

5.2 Routing Engine Configuration

NOTE: For all the examples in this document going forward the term 'inbound call' refers to a call in the TDM to IP direction and the term 'outbound call' refers to a call in the IP to TDM direction.

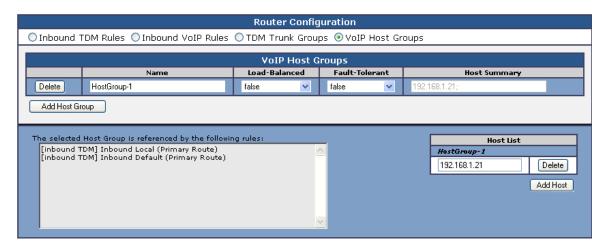
The example given in the system diagram at the start of this integration guide has the following dialing plans in the system:

- All TDM side stations have DID numbers assigned in the 2xxx extension range.
- All OCS side stations have DID numbers assigned in the 5xxx extension range.

All inbound calls need to be sent through to the Mediation Server at a specific IP address.

5.2.1 VolP Host Group configuration

The first item to take care of is to set up the IP endpoint to use as the IP destination for all inbound calls. This is done in the routing table under the section VoIP Host Groups. Define a single host group (using the default group is fine) that includes the IP address of the gateway listening side of the Mediation Server; in this example case the IP address 192.168.1.21 is for this.



5.2.2 TDM and VoIP Routing Rule Configuration

The second item is to configure the routing rules that will associate inbound or outbound calls with the proper digit manipulation rules for the type of call they need to service. This will require that the gateway perform some digit manipulation on calls that go from the TDM side to the IP side as well as in the reverse direction, IP to TDM.

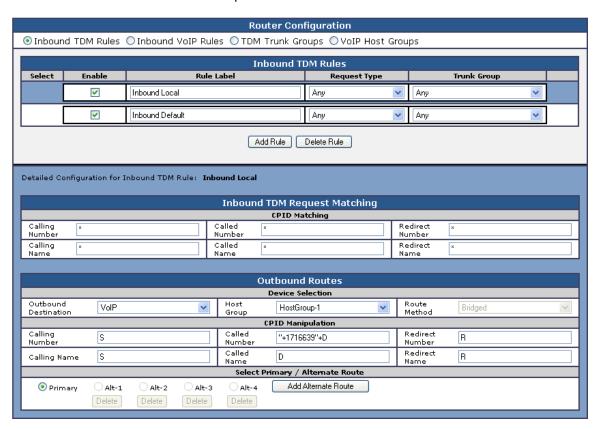
The major idea here to remember is that OCS expects to get, and will send out, all addresses in E.164 format. This means that the gateway needs to recognize the need to convert up and down as needed to and from this format as calls pass through. To do this you make use of the Routing engine's CPID manipulation rules.

5.2.2.1 Inbound TDM Rules

When a local user on the PBX picks up their phone and calls one of the extensions on the OCS side within the 5xxx range the gateway will receive a call with a calling party of 4 digits. It then needs to convert that number up to full E.164 format and send the call on to OCS.

This example will take any number and then convert it into the full E.164 format by concatenating a prefix of '+1716639' onto the front of the number where 716 is the area code and 639 is the local exchange.

Other calls, such as DIDs that arrive over TDM trunks from the PSTN may provide a full 10 digits to the PBX or they may only provide the extension number after the prefix has been stripped off by the PBX. Depending on your site specific requirements you may need to add or build different rules to handle these cases. An example of the inbound rule for local PBX users is shown below:



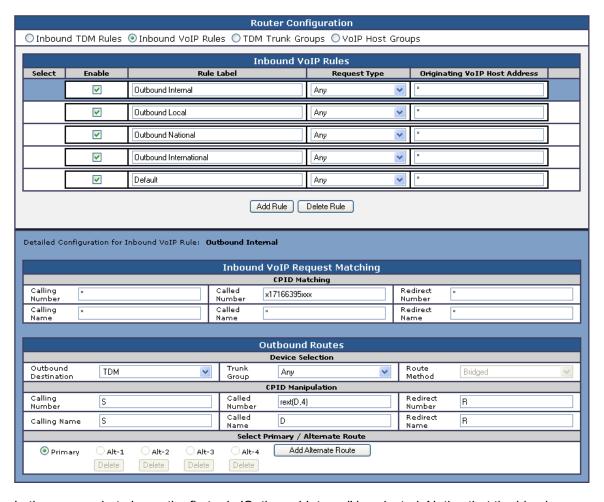
The CPID matching rule is simply a * meaning that any dialed number from a local user presented to this trunk will be seen by this rule. The CPID manipulation rule then uses the digits that are being seen (in this example it will be a 4 digit number because that is how the trunk is programmed) and then adds the prefix of "+1716639" onto it to build the full E.164 number that is needed for OCS. This rule also sets the destination to the VoIP Host group defined previously that points to the inbound IP address of the Mediation Server.

In addition to this rule a default rule has been left in place that acts as a catch all. This rule performs no CPID manipulation at all and just tries to send the call to the VoIP host group as dialed.

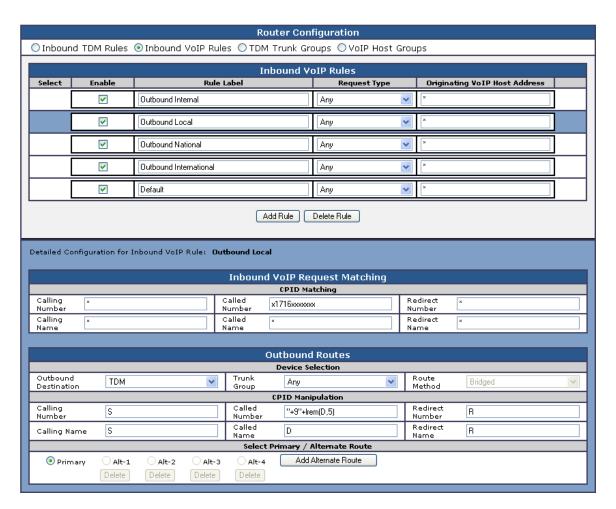
5.2.2.2 Inbound VoIP Rules

When an OCS user dials a number OCS will, through the use of normalization rules in the Location profile, provide the gateway with a number in full E.164 format. The gateway needs to be able to recognize various number patterns in inbound IP calls and properly manipulate them for the outbound TDM call that results.

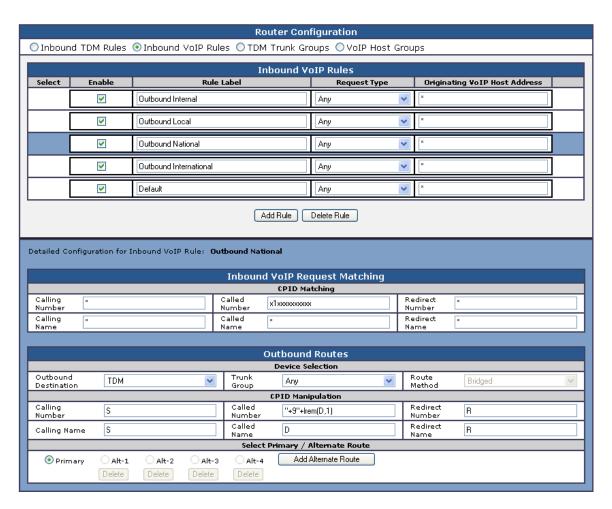
In the example here, OCS has been setup (as you will see later) with a route that directs all calls that meet the pattern 5xxx to the gateway in full E.164 format. The gateway then needs to know how to identify these numbers as extensions that are local on the PBX and manipulate them accordingly. To do this it needs to simply extract the right 4 digits from the called number provided to remove the prefix of "+1716639" and leave the last 4 digits remaining. Local, national and international numbers are going to need to be manipulated. At very least they will need a trunk access number, like a 9, pre-pended onto the front of them in order to dial an outside line. These can also be done using manipulation rules as follows:



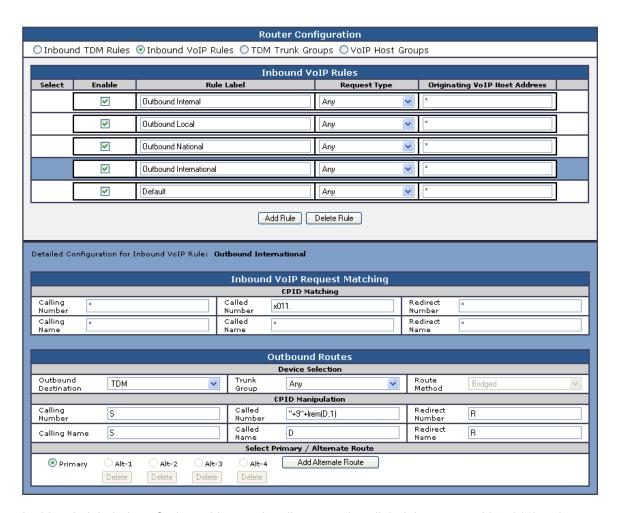
In the screen shot above, the first rule 'Outbound Internal' is selected. Notice that the blue bar near the top of the screen highlights this rule. The lower half of the screen displays the details of the currently selected rule. This rule matches outbound calls that have a called party number that starts with '+17166395' followed by any three digits. This rule is designed to match the locally defined TDM extensions as shown in the first figure in this document. Calls that match this rule are meant to go to a local user on the PBX. The CPID manipulation section of this rule extracts the last four digits from the called party number. The extracted four digits are then dialed as a local extension on the PBX.



In the screen shot above, the rule 'Outbound Local' is selected. This rule matches outbound calls that have a called party number that starts with '+1716' followed by seven digits. This rule is designed to match the calls within the same area code, but not from the same PBX. Calls that match this rule are meant to go to a local user that is not on the PBX. In the CPID manipulation area the trunk access code is added to the string and the leading 5 characters are stripped off (the '+1716'). The full string out as '+9xxxxxxxx' is sent.



In this rule labeled as 'Outbound National' any number dialed that starts with '+1' and includes 10 digits indicates a number that is not in the local area code. In this case the CPID manipulation simply adds a +9 to the start of the number and strips off the leading + creating a result of '+91xxxxxxxxxx'.



In this rule labeled as 'Outbound International' any number dialed that starts with '+011' and includes any number of digits indicates a number that is not in the local area code. In this case the CPID manipulation simply adds a +9 to the start of the number and strips off the leading + creating a result of '+9011xxxxxxxxxxx'.

The last rule that is defined is another default rule that acts as a catch all and simply attempts to dial any number provided that has not matched the previous rules in the list.

Note 1: The last two rules labeled as 'Outbound National' and 'Outbound International' COULD have been combined into one rule since the CPID manipulation was the same in both. The rules have been split out here in this example simply for clarity of the example. Also, if the environment uses different trunks for local, national (long Distance) and international calls, breaking these rules out into separate segments allows you to also define trunk groups and direct calls of these specific types to those individual trunks.

Note 2: The rules are evaluated in the order they are listed, top down. The first rule that matches is used so the order is important. Always consider placing your more specific rules at the top of the order and the more general at the bottom.

6. PBX Setup Notes

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

- Activating the QSIG protocol.
- Administrating the Trunk configuration.
- Assigning an access code to the trunk.
- Setting up the subscribers stations.

Programming on the Hipath 4000 can be done either by using a text based GUI application or a command line interface. The programming here will be shown using the GUI commands.

6.1 Activating QSIG Protocol

If required you may need to turn on the QSIG protocol before use. The programming steps below show how it was done on the test system. If you have any questions we recommend that you contact your Siemens representative.

Enter the DIS-PRODE: DB, PDSHORT; command and press RETURN. The following screen is displayed:

Make note of the number shown in the PDNAME column. You will need it in a future step.

Enter the DISPLAY-PRODE: SRC=HD, KIND=PDSHORT; command and press RETURN. The following screen will be displayed:

```
+----+
| PDNO | PDSTRING | IDENT | VERSION |
+----+
| 32 | ISO QSIG Second ed. SS | H/08 | B0-EL0.20.001 |
+----+
```

Make note of the number shown in the PDNO column. You will need it in a future step.

Enter the REG-PRODE; command and press RETURN.

Enter the COPY-PRODE: PD, 32, PD14; command and press RETURN.

Enter the CHANGE-PRODE: VARTAB, PSS1V2R, PD14, ,; command and press RETURN. The following screen should be displayed:

```
AMO-PRODE-111 PROTOCOL DESCRIPTOR FOR NETWORKING REGENERATE COMPLETED;
```

6.2 Administering Trunk Configuration

Use the add-buend command to configure a trunk group.

- Enter the add-buend command and press RETURN
- At the prompt TGRP = enter xxx
 - o where xxx is any available trunk group number and press RETURN
- At the prompt NAME = enter xxx
 - where xxx is any assigned name for the trunk group and press RETURN
- At the prompt NO = enter 48 and press RETURN
- At the prompt TRACENO = enter 0 and press RETURN
- At the prompt ACDTHRH = enter * and press RETURN
- At the prompt PRIONO = enter 2 and press RETURN
- At the prompt TDDRFLAG = enter OFF and press RETURN
- At the prompt GDTRRULE = enter 0 and press RETURN
- At the prompt ACDPMGRP = enter 0 and press RETURN
- At the prompt CHARCON = enter NEUTRAL and press RETURN

Display and validate the changes with the following command:

Enter DIS-BUEND: 100; and press RETURN. The following screen should be displayed:

Use the add-tdcsu command to configure a trunk.

- Enter the add-tdcsu and press RETURN
- At the prompt OPT = enter new and press RETURN
- At the prompt PEN = enter x-xx-xxx-x
 - \circ Where x-xx-xxx-x is the location of the installed T1 Port Equipment Number and press <code>RETURN</code>
- At the prompt COTNO = enter xxx
 - O Where XXX is your selected class of trunk and press RETURN
- At the prompt COPNO = enter xxx
 - o Where xxx is your selected class of parameter and press RETURN
- At the prompt DPLN = enter 0 and press RETURN
- At the prompt ITR = enter 0 and press RETURN
- At the prompt COS = enter 100 and press RETURN
- At the prompt LCOSV = enter 1 and press RETURN
- At the prompt LCOSD = enter 5 and press RETURN
- At the prompt CCT = enter CORNET-NQ and press RETURN
- At the prompt DESTNO = enter 100 and press RETURN
- At the prompt PROTVAR = enter PSS1V2R and press RETURN
 - This is the setting that specifies that you are using QSIG with supplemental services. The protocol you have specified is 'qsig iso iso/iec 11572 2nd. with ss'
- At the prompt SEGMENT = enter 1 and press RETURN
- At the prompt DEDSCC = and press RETURN
- At the prompt DEDSVC = enter NONE and press RETURN
- At the prompt FACILITY = and press RETURN
- At the prompt DITIDX = and press RETURN
- At the prompt SRTIDX = and press RETURN
- At the prompt TRTBL = enter GDTR and press RETURN
- At the prompt SIDANI = enter N and press RETURN
- At the prompt ATNTYP = enter TIE and press RETURN
- At the prompt CBMATTR = enter NONE and press RETURN
- At the prompt TCHARG = enter N and press RETURN

- At the prompt SUPPRESS = enter 0 and press RETURN
- At the prompt DGTPR = and press RETURN
- At the prompt ISDNIP = and press RETURN
- At the prompt ISDNNP = and press RETURN
- At the prompt PNPL2P = and press RETURN
- At the prompt PNPL1P = and press RETURN
- At the prompt PNPAC = and press RETURN
- At the prompt TRACOUNT = enter 31 and press RETURN
- At the prompt SATCOUNT = enter MANY and press RETURN
- At the prompt NNO = enter 1-1-100 and press RETURN
- At the prompt ALARMNO = enter 0 and press RETURN
- At the prompt FIDX = enter 1 and press RETURN
- At the prompt CARRIER = enter 1 and press RETURN
- At the prompt ZONE = enter EMPTY and press RETURN
- At the prompt COTX = enter 100 and press RETURN
- At the prompt FWDX = enter 5 and press RETURN
- At the prompt CHIMAP = enter N and press RETURN
- At the prompt INIGHT = and press RETURN
- At the prompt DOMTYPE = enter PRIVATE and press RETURN
- At the prompt DOMAINNO = enter 0 and press RETURN
- At the prompt TPROFNO = and press RETURN
- At the prompt CCHDL = and press RETURN
- At the prompt UUSCCX = enter 16 and press RETURN
- At the prompt UUSCCY = enter 8 and press RETURN
- At the prompt FNIDX = enter 0 and press RETURN
- At the prompt NWMUXTIM = enter 10 and press RETURN
- At the prompt SRCGRP = enter 1 and press RETURN
- At the prompt CLASSMRK = enter EC&G711 and press RETURN
- At the prompt TCCID = and press RETURN
- At the prompt TGRP = enter 100 and press RETURN
- At the prompt SRCHMODE = enter DSC and press RETURN
- At the prompt INS = enter Y and press RETURN
- At the prompt DEV = enter S1CONN and press RETURN
- At the prompt BCHAN = enter 1 & & 23 and press RETURN
- At the prompt BCNEG = enter N and press RETURN
- At the prompt BCGR = enter 1 and press RETURN
- At the prompt LWPAR = enter 100 and press RETURN
- At the prompt LWPP = and press RETURN
- At the prompt LWLT = and press RETURN
- At the prompt LWPS = and press RETURN
- At the prompt LWR1 = and press RETURN
- At the prompt LWR2 = and press RETURN
- At the prompt SVCDOM = and press RETURN

Once completed you can validate the settings by using the <code>DIS-TDCSU:<pen></code> command where the PEN is your Peripheral Equipment Number of your trunk card. Press enter and the following screen should be displayed.

Use the add-cot to adjust the class of trunk setting (COTNO) that you are using in the trunk configuration.

- Enter the add-cot command and press RETURN
- At the prompt COTNO = enter xxx
 - Where xxx is the class of trunk (COTNO) that you have used in the trunk configuration using the add-tdcsu command above and press RETURN
- At the prompt PAR = press RETURN
- At the prompt DEV = enter s1conn and press RETURN
- At the prompt INFO = press RETURN

Repeat these steps for each of these parameters

PRI, RCL, XFER, KNOR, CEBC&CBBN&CBFN&IEVT&BSHT&BLOC&PROV&ATRS&ROPT&TSCS&TRSC&C&CFOS&PINR&AOCC&CTLS&AMFC&NTON;

Once completed you can display your class of trunk configuration using the DISP-COT: 100, L, , , , , ; command. You should see a screen that looks like the following:

```
COT: 100 INFO:
  DEVICE: INDEP
                       SOURCE: DB
  PARAMETER:
        PRIORITY FOR AC WILL BE DETERMINED FROM MESSAGE
                                                                    PRT
       RECALL IF USER HANGS UP IN CONSULTATION CALL
                                                                    RCT.
       TRUNK CALL TRANSFER
                                                                    XFER
       KNOCKING OVERRIDE POSSIBLE
                                                                    KNOR
       CALL EXTEND FOR BUSY, RING OR CALL STATE
                                                                    CEBC
       NETWORKWIDE AUTOMATIC CALLBACK ON BUSY
                                                                    CBBN
       NETWORKWIDE AUTOMATIC CALLBACK ON FREE
                                                                    CBFN
       REGISTRATION OF IMPLAUSIBLE EVENTS
                                                                    TEVT
       DON'T RELEASE CALL TO BUSY HUNT GROUP
                                                                    BSHT
       END-OF-DIAL FOR BLOCK IS SET
                                                                    BLOC
       EMERGENCY OVERRIDE/DISCONNECT VIA S0/S2 LINE
                                                                    PROV
       ACTIVATE TRANSIT COUNTER ADMINISTRATION FOR S0/S2 LINE
                                                                    ATRS
        CONNECTION TO ROUTE OPTIMIZATION NODE
                                                                    ROPT
       TSC-SIGNALING FOR NETWORKWIDE FEATURES (MANDATORY)
                                                                    TSCS
       TRUNK SENDS CALL CHARGES TO ORIGINATING NODE NUMBER
                                                                    TRSC
       CALL FORWARDING PROGRAMING FOR OTHER SUBSCRIBERS
                                                                    CFOS
       PIN NETWORKWIDE POSSIBLE
                                                                    PINR
       AOC PER CALL (AUTOMATICAL OR ON REQUEST), MAND. CORNET-NQ
                                                                    AOCC
       CONTROLLED TRUNK AND LINE SELECTION
                                                                    CTLS
                                                                    AMFC
        AUTOM.DTMF CONVERSION ON INCOM.CALL WHILE IN TALK STATE
       NO TONE
                                                                    NTON
AMO-COT -111
                   CLASS OF TRUNK FOR CALL PROCESSING
DISPLAY COMPLETED;
```

Use the add-cop command to configure the class of parameter setting (COPNO) that you are using in the trunk configuration.

- Enter the add-cop command and press RETURN
- At the prompt COPNO = enter xxx
 - o Where xxx is the class of parameter (COPNO) that you have used in the trunk configuration using the add-tdcsu command above and press RETURN
- At the prompt PAR = enter L3AR and press RETURN
- At the prompt TRK = press RETURN
- At the prompt TOLL = press RETURN
- At the prompt DEV = press RETURN
- At the prompt INFO = press RETURN

Repeat these steps for each of these parameters

LKNQ, RRST

Once completed you can display your class of parameter configuration using the DISP-COP: 100, L, , , , , ; command. You should see a screen that looks like the following:

```
H500: AMO COP STARTED

COP: 100 INFO:
DEVICE: INDEP SOURCE: DB
PARAMETER:
REGISTRATION OF LAYER 3 ADVISORIES
LINK OF 2 CORNET-NQ PABX VIA INTER-LINK
REFLECT RESTART INDICATOR AND B-CHANNEL BY RESTART

AMO-COP -111 CLASS OF PARAMETER FOR DEVICE HANDLER
DISPLAY COMPLETED;
```

Shown below is an example of the trunk routing set up on a switch using a T1 QSIG trunk. This is to be used as an example only, as many parameters will be site specific and should be configured by a vendor technician.

	FOR ALL DPLN						SV	C = V(CE
CODE	+	TGRP CCNO 	P L+- B C	+: NV DS	DTMF -+	ا ++	LRTE	CPAR 	F
 20 NEUTRAI	12345 67890 123452 	+ 100 	+ 1	+	+	PAUSE 		+	B +
ROUTOE DTMFCN DTMFPU EMCYRT INFO = NOPRCE NITO = CLNAME	WD = NO : NO :DL = NO :::100 LDAT QSIG	YES = WIT IN RO NE = N	PL: 'HOU' OUTA'	B = NG I DTM IT = 1 RERI I	FWDBL = NO FTEXT = NO MAINGRP = NGRP = NO RTENO : CUTRAL) SUBGROUP :	= 4	 		
LRTE = 101 NAME = NI2 (NEUTRAL) LSVC = ALL DNNO = 101 PDNNO = 0 DESTNO = 101 ROUTOPT = YES REROUT = YES PLB = NO FWDBL = NO DTMFCNV = FIX DTMFDSP = WITHOUT DTMFTEXT = DTMFPULS = PP80 BUGS = LIN ROUTATT = NO MAINGRP = 5 EMCYRTT = NO CONFTONE = NO RERINGRP = NO RTENO = 5 INFO = NOPRCFWD = NO NITO = NO CLNAMEDL = NO									
DNNO = ROUTOF DTMFCN DTMFPU EMCYRT INFO = NOPRCF NITO =	IV = FIX DTMFDSP ILS = PP80 BUGS = L: TT = NO CONFTOI : WD = NO :: NO EDL = NO	= WIT	UTA'	TT = 1	NO MAINGRP =				

In this example we se that a route has been set up to take calls delivered to route # 20 (as shown in the CODE field) and send them to trunk group (TRGP) 100 that we have defined in the prior steps.

6.3 Accessing Code Assigned to the Trunk

Shown below is how an access code is assigned to the trunk routing.

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS					
l CODE	Ì	PROGRESS STATE 1 11111 11112 22 7890 12345 67890 12	ANALYSIS	DNI/ADD-	-INFO	 IRT 	
3002 	*** * 	**** ***		DESTNO 2 DNNO (PDNNO (0-0-		
		LANS, FEATURE ACCESS	3 CODES				
SPLAY COMPLET: S-RICHT:PM;				1			
S-RICHT:PM;		+		TYPE	-+		

In this example we see that the access code 3002 has been assigned to route calls made to it to trunk route 20 as defined in the DESTNO field. This setup allows both subscribers to call the trunk access code 3002 and get through to the gateway as well as allow subscribers station sets to be forwarded o the access code for converge under busy and ring no answer conditions.

6.4 Setting Up Subscriber Station Sets

There is no PBX side programming for setting up the subscriber station sets. All the forwarding of the subscriber station sets is defined directly on subscriber station set using either feature access codes or the phones soft menu keys. Subscribers should be directed to set their internal and external ring no answer and busy forwarding conditions to the extension number assigned to the access code assigned to the trunk route.

6.5 Additional Comments

- Phonemail access must be configured in RICHT (Parameter PM)
- Index of RICHT: PM must be setup for every extension that needs access (SBCSU => Parameter PMIDX)
- COS of the extension must contain TTT (Trunk to Trunk Transfer) and FWDEXT (ForWarDdingEXTernal / AMO COSSU) to forward to the server
- Make sure you don't have CFVA set for the Trunk the server is connected to (AMO COT This parameter will check the availability of the forwarding target)

7. Microsoft OCS setup

7.1 Steps for configuring OCS

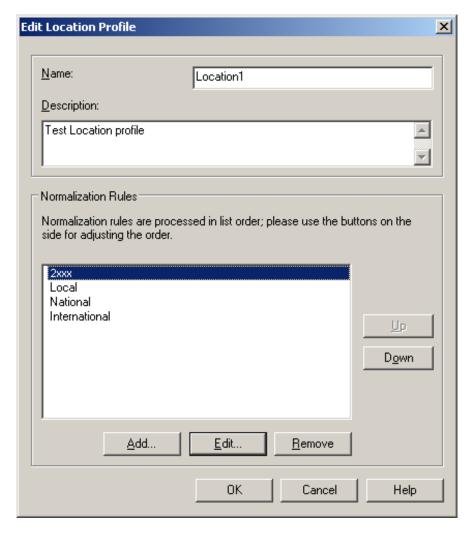
Normalization rules are used to convert all possible dial numbers into full E.164 formatted numbers. Microsoft OCS uses the standard E.164 format to search for all users listed in Active Directory (AD).

When an 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 that are for internal extensions, local numbers, long distance numbers, and international numbers.

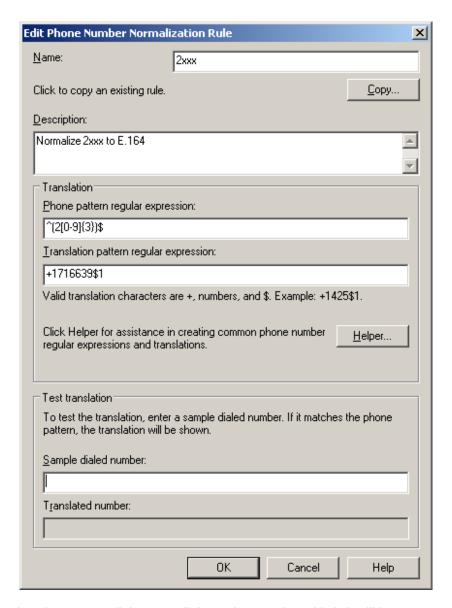
From the Start menu select the following to configure the OCS server:

Programs → Administrative Tools → OCS 2007

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



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



In this example, when a user dials any 4-digit number starting with 2, it will be converted to its E.164 equivalent of +1716639xxxx and then that number will be searched for in AD.

More examples are shown in the following table:

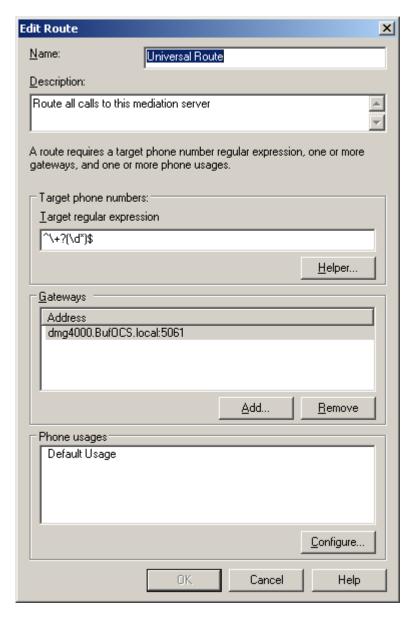
Name	Phone Pattern	Translation Pattern	Descriptions
2xxx	^(2[0-9]{3})\$	+1716639\$1	Normalize 2xxx to E.164
Local	^(\d{7})\$	+1716\$1	Local number
National	^1(\d*)\$	+1\$1	Long distance number
International	^011(\d*)	+011\$1	International number

A default route is used to route all calls to the Mediation server. If you need to route some calls to a different Mediation server, configure the Target phone numbers field accordingly.

From the Start menu select the following to configure the OCS server:

Programs → Administrative Tools → OCS 2007

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



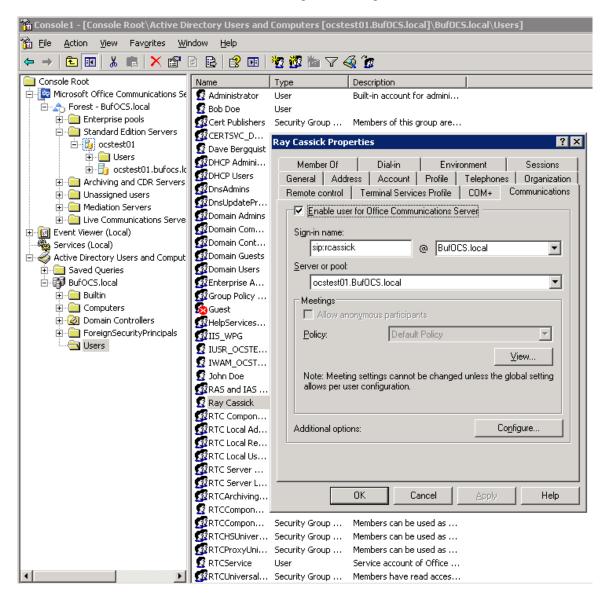
This entry routes any number with or without '+' prefix followed by any digits to Mediation server dmg4000.bufocs.local

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

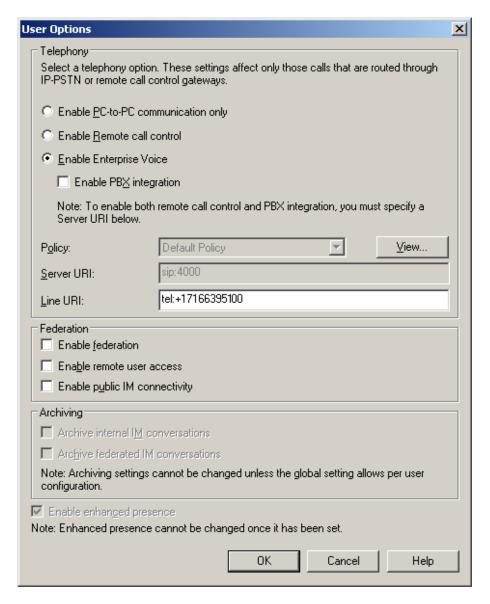
Note: Unless the dialed number from OCS client (such as Office Communicator) is in E.164 format, OCS must find a normalization rule to convert the dialed number to E.164.

7.2 Steps for configuring OCS clients

The domain users need to be enabled for making calls through OCS server.



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



In the above configuration for user Ray Cassick, when an inbound PSTN call for 5100, it will be converted by the gateway CPID manipulation and routing rules into +17166395100. OCS will match that number provided by the gateway to the Line URI parameter for this user and ring Ray Cassick if he is logged on to OCS from Office Communicator or any OCS supported device.

8. Testing Validation Matrix

The table below shows various test scenarios that are run as typical validation scenarios when the 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 complete sample showing call flows and states please consult the Gateway SIP Compatibility Guide.

Test Number	Call Scenario Description	Notes
Inbound ca	II scenarios	
1	Direct call from TDM station set to OCS client.	
2	Direct call from OCS client to TDM station set.	

9. Troubleshooting

9.1 Important Debugging Tools

- Ethereal/Wireshark Used to view and analyze the network captures provided by the Dialogic gateway diagnostic firmware.
- Adobe Audition -- Used to review and analyze the audio extracted from the network captures to troubleshoot any audio related issues.

9.2 Important Gateway Trace Masks

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

- voip prot and voip code this allows the collection of all SIP related messages as
 they are sent from and received by the gateway. This data is important in cases where
 you feel that the gateway is not able to communicate properly with the messaging server.
- tel event and tel code This allows the collection of all circuit side activity of the emulated station set such as display updates, key presses, light transitions and hook state changes. This data is very important in the following scenarios:
 - o Call control problems (dropped calls, failing transfers, etc...)
 - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc...)
- teldrv prot This allows the collection of all ISDN messages both transmitted and received on the gateways front end interface. This data is very important in the following scenarios:
 - Call control problems (dropped calls, failing transfers, etc...)
 - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc...)
- Routingtable (all keys) This allows you to look inside the routing table engine
 and see how matching rules and CPID manipulation rules work with respect to your call.
 This data is very important in the following scenarios:

o Call routing problem (reaching the incorrect OCS client or no client at all, etc...)

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

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