

### 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

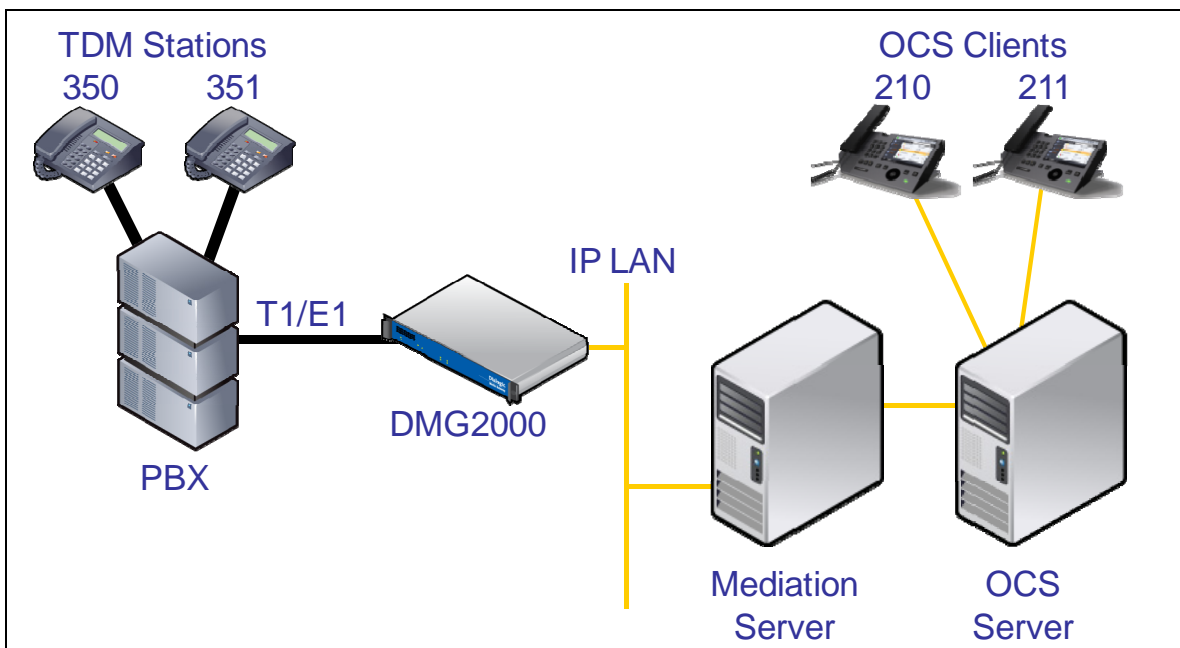
PBX Vendor	Mitel
Model	3300 ICP
Software Version	7.1.6.13
Additional Notes	N/A

#### 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.

[http://www.mitel.com/resources/392\\_891-51008713RI-EN\\_FINAL.pdf](http://www.mitel.com/resources/392_891-51008713RI-EN_FINAL.pdf)

##### **3.1.1 PBX Equipment Required**

To support the T1 QSIG configuration as documented you need a Mitel 3300 Universal NSU (50001270) with either one or two T1 modules in it. The NSU is external to the 3300 switch.

The 3300 switch must have a fiber optical interface module (typically placed in Module bay number 1).

##### **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.

Cabling from the NSU to the switch is done with fiber optical cable (FIM link).

#### **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:

- 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	

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

T1/E1 Port Selection	
Select Port to Modify	all ports

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

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

Voip General Settings	
User-Agent	
* Host and Domain Name	pbxgw.default.com
Transport Type	TCP
Call as Domain Name?	No
SIPS URI Scheme Enabled	No
Invite Expiration (sec)	120

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

VoIP Media Settings		
Audio		
* Audio Compression	G.711u/G.711a	
RTP Digit Relay Mode	RFC2833	
RTP Fax/Modem Tone Relay Mode	Inband-Tone	
* RTP Source IP Address Validation	Off	
* RTP Source UDP Port Validation	Off	
Signaling Digit Relay Mode	Off	
Voice Activity Detection	Off	
RFC 3960 Early 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.*

In the example given in the system diagram at the start of this integration guide we see that we have the following dialing plans in the system:

- All TDM side stations have DID numbers assigned in the 3xx extension range.
- All OCS side stations have DID numbers assigned in the 2xx extension range.

We also know that we need to send all inbound calls through to the Mediation Server at a specific IP address.

### 5.2.1 VoIP Host Group configuration

The first item we should take care of is to set up our IP endpoint to use as our IP destination for all our inbound calls. This is done in the routing table under the section VoIP Host Groups. We 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 our example case we are using the IP address 192.168.1.21 for this.

**Router Configuration**

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

**VoIP Host Groups**

	Name	Load-Balanced	Fault-Tolerant	Host Summary
Delete	HostGroup-1	false	false	192.168.1.21;

The selected Host Group is referenced by the following rules:

[Inbound TDM] Inbound Local (Primary Route)  
 [Inbound TDM] Inbound Default (Primary Route)

**Host List**

HostGroup-1
192.168.1.21

## 5.2.2 TDM and VoIP Routing Rule Configuration

The second item we need to configure are 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 2xx range the gateway will receive a call with a calling party of 3 digits. It then needs to convert that number up to full E.164 format and send the call on to OCS.

In our example here we need to take any number that starts in the 2xx range and then convert it into the full E.164 format by concatenating a prefix of '+17166393' onto the front of the number where 716 is the area code and 639 is the local exchange.

Other calls, such as DID's 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. In our example the inbound rule we use for local PBX users is shown below:

**Router Configuration**

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

Select	Enable	Rule Label	Request Type	Trunk Group
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Inbound Local	Any	Any
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Inbound Default	Any	Any

Detailed Configuration for Inbound TDM Rule: **Inbound Local**

Inbound TDM Request Matching			
CPID Matching			
Calling Number	*	Called Number	*
Calling Name	*	Called Name	*
Redirect Number	*	Redirect Name	*

Outbound Routes		
Device Selection		
Outbound Destination	Host Group	Route Method
VoIP	HostGroup-1	Bridged
CPID Manipulation		
Calling Number	Called Number	Redirect Number
S	"*+17166393"+D	R
Calling Name	Called Name	Redirect Name
S	D	R
Select Primary / Alternate Route		
<input checked="" type="radio"/> Primary <input type="radio"/> Alt-1 <input type="radio"/> Alt-2 <input type="radio"/> Alt-3 <input type="radio"/> Alt-4		
<input type="button" value="Add Alternate Route"/>		
<input type="button" value="Delete"/> <input type="button" value="Delete"/> <input type="button" value="Delete"/> <input type="button" value="Delete"/>		

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 case it will be a three digit number starting with 2 because that is how the trunk is programmed) and then adds the prefix of "+17166393" 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 we have defined previously that points to the inbound IP address of the Mediation Server.

In addition to this rule, there is a default rule left in place that acts as a catch all. This rule does not do CPID manipulation at all and just sends 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 so 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 our example here, OCS has been setup (as you will see later) with a route that directs all calls that meet the pattern 5xx 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 3 digits from the called number provided and removing the prefix of "+17166393" (see the CPID Manipulation section of the next screen shot below).

**Router Configuration**

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

Inbound VoIP Rules				
Select	Enable	Rule Label	Request Type	Originating VoIP Host Address
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound Internal	Any	*
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound Local	Any	*
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound National	Any	*
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound International	Any	*
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Default	Any	*

Detailed Configuration for Inbound VoIP Rule: **Outbound Internal**

Inbound VoIP Request Matching			
CPID Matching			
Calling Number	*	Called Number	x17166393xxx
Calling Name	*	Called Name	*
Redirect Number	*	Redirect Name	*

Outbound Routes		
Device Selection		
Outbound Destination	TDM	Trunk Group
	Any	Route Method
	Bridged	
CPID Manipulation		
Calling Number	S	Called Number
	5	next(D,3)
Calling Name	S	Called Name
	5	D
Redirect Number	R	Redirect Name
	R	R
Select Primary / Alternate Route		
<input checked="" type="radio"/> Primary <input type="radio"/> Alt-1 <input type="radio"/> Alt-2 <input type="radio"/> Alt-3 <input type="radio"/> Alt-4 <input type="button" value="Add Alternate Route"/>		
<input type="button" value="Delete"/> <input type="button" value="Delete"/> <input type="button" value="Delete"/> <input type="button" value="Delete"/>		

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 (meaning VoIP to TDM) that have a called party number that starts with '+17166393' 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 three digits from the called party number. The extracted three digits are then dialed as a local extension on the PBX.

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, prepended onto the front of them in order to dial an outside line. Rules to do this kind of manipulation are shown in the examples below.

**Router Configuration**

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

**Inbound VoIP Rules**

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

**Detailed Configuration for Inbound VoIP Rule: Outbound Local**

**Inbound VoIP Request Matching**

**CPID Matching**

Calling Number	x	Called Number	x1716xxxxxxx	Redirect Number	*
Calling Name	x	Called Name	x	Redirect Name	*

**Outbound Routes**

**Device Selection**

Outbound Destination	TDM	Trunk Group	Any	Route Method	Bridged
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**CPID Manipulation**

Calling Number	S	Called Number	"*+9"+lrem(D,5)	Redirect Number	R
Calling Name	S	Called Name	D	Redirect Name	R

**Select Primary / Alternate Route**

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

This rule, labeled as 'Outbound Local', matches any number that starts with '+1716' followed by seven digits. This indicates that it is a local number (outside of the PBX but within the local calling area code of 716) and does not need the area code dialed as part of the number. The CPID manipulation section for the Called Number adds a trunk access code of '+9' to the front of the string and lrem(D,5) strips off the leading five characters (the '+1716'). This sends the full string out as '+9xxxxxxx', meaning '+9' followed by the last seven digits of the number from OCS.

**Router Configuration**

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

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**Inbound VoIP Rules**

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

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Detailed Configuration for Inbound VoIP Rule: **Outbound National**

**Inbound VoIP Request Matching**

CPID Matching			
Calling Number	*	Called Number	x1xxxxxxxx
Calling Name	*	Called Name	*

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**Outbound Routes**

Device Selection		
Outbound Destination	TDM	Trunk Group
Route Method	Bridged	

CPID Manipulation		
Calling Number	S	Called Number
Calling Name	S	Called Name

**Select Primary / Alternate Route**

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

This rule, labeled as 'Outbound National', matches any number dialed that starts with '+1' followed by ten digits. This will only match numbers not in the local area code because calls to the local area code were handled by the previous rule 'Outbound Local'. The CPID manipulation section strips off the leading '+', add '+9' to the start of the number creating a result of '+91xxxxxxxx'.



**Router Configuration**

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

Inbound VoIP Rules				
Select	Enable	Rule Label	Request Type	Originating VoIP Host Address
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound Internal	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound Local	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound National	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Outbound International	Any	*
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Default	Any	*

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Detailed Configuration for Inbound VoIP Rule: **Outbound International**

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*	Called Number	x011.	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes			
Device Selection			
Outbound Destination	TDM	Trunk Group	Any
Route Method	Bridged		

CPID Manipulation			
Calling Number	S	Called Number	"*9"+rem(D,1)
Redirect Number	R		
Calling Name	S	Called Name	D
Redirect Name	R		

**Select Primary / Alternate Route**

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

This rule, labeled as 'Outbound International', matches any that starts with '+011' and includes any number of digits after that indicating a number that is not in our local area code and is indeed an international number. In this case the CPID manipulation adds a '+9' to the start of the number and strips off the leading '+' creating a result of '+9011xxxxxxxx'.

The last rule that you see 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. We have split them out here in this example simply for clarity of the example. Also, if your 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 messaging system are as follows:

- Configuring hardware and class of service.
- Configuring ISDN-PRI interface.
- Configuring ARS options.
- Setting up subscriber station sets.

All PBX programming is done via a web browser by connecting to the network port of the PBX.

### 6.1 Configure Hardware and Class of Service

Use the *Units and Modules Network Services Unit Configuration* menu selection to configure a Network Service Unit. This command sets the options on an installed NSU so it can be configured for a specific type and protocol to be used with the gateway. Select an NSU from the list and click the Change button to configure the options.

The screenshot shows the web interface for the Mitel Networks 3300 Integrated Communications Platform (ICP). The browser title is "iPBX3 - Mitel Networks® 3300 Integrated Communications Platform (ICP) - Microsoft Internet Explorer". The page displays an "Alarm Status: Critical 2006-Jul-27 10:01:13" and a "System Message" section. The main content area is titled "Network Services Unit Configuration" and contains a table with the following data:

Unit	Unit Type	Controller Module	Port	Programmed Type	Protocol	Comment
6	Primary Network Services Unit	1	1	3300 Universal NSU	Universal T1	
7	Primary Network Services Unit	1	2	3300 Universal NSU	Universal E1	
8						
9						
10						
11						
12						
13						
14						

The interface also includes a navigation tree on the left with categories like System Configuration, System Capacity, Units/Modules, and Trunks. At the bottom, the Mitel logo and "3300 ICP" are visible.

See the screen below for the selected options.

-- Web Page Dialog

### Network Services Unit Configuration

Unit: 6

Unit Type: Primary Network Services Unit

Controller Module: 1

Port:  1  2

Programmed Type: 3300 Universal NSU

Protocol: Universal T1

Comment:

Save Cancel

Before moving on to the next task be sure to click the Save button.

Configure a Class of Service Template for the Trunk. Shown below is the listing of all the enabled and disabled classes of service that will be configured on the trunk interface in this example. Yours may vary depending on site requirements but keep in mind that disabling certain classes of service will have an effect on certain available features.

-- Web Page Dialog

### Class of Service Options Assignment

Class Of Service Number: 1

Comment: Radbrook Config

Account Code Verified:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ACD Silent Monitor Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ACD Silent Monitor Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ACD Silent Monitor Notification:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ANI/DNIS/ISDN Number Delivery Trunk:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Auto Answer Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Autovon Auto-preemption:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Autovon Trunk:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Brokers Call:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Busy Override Security:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Call Announce Line:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forwarding Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forwarding (External Destination):	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forwarding (Internal Destination):	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forward Override:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Call Hold:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Hold Remote Retrieve:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Hold - Retrieve with Hold Key:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Pickup Dialed Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Pickup Directed Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Privacy:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Call Reroute after CFFM to Busy Destination:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Waiting Swap:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Calling Name Display - Internal - ONS:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Calling Number Display - Internal - ONS:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Campon Tone Security:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Check COR after PSTN Dial Tone:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Clear All Features Remote:	<input checked="" type="radio"/> No	<input type="radio"/> Yes

Conference Call:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
COV/ONS/E&M Voice Mail Port:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
DASS II OLI/TLI Provided:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Dialled Night Service:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Disable Send Message:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Display ANI/ISDN Calling Number Only:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Display ANI/DNIS/ISDN Calling/Called Number:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display Caller ID on multicall/keylines:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display DNIS/Called Number Before Digit Modification:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display Dialed Digits during Outgoing Calls:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display Held Call ID on Transfer:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Do Not Disturb:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Do Not Disturb - Access to Remote Phones:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Do Not Disturb Permanent:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Emergency Call Notification - Audio:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Emergency Call Notification - Visual:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Enable Call Duration Limit on External Calls:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Enable Call Duration Limit on Internal Calls:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Executive Busy Override:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
External Trunk Standard Ringback:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Flexible Answer Point:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Forced Verified Account Code:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Forced Non-Verified Account Code:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Call Forward Follow Me Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Call Forward Follow Me Allow:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Page Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Page Allow:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Handset Volume Adjustment Saved:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Handsfree AnswerBack Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
HCI/CTI/TAPI Call Control Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes

<b>HCI/CTI/TAPI Monitor Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Head Set Switch Mute:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hot Desk Remote Logout Enabled:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hot Desk Login Accept:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hotel Room Extension:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hotel Room Monitor Setup Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hotel Room Monitoring Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hotel/Motel Room Personal Wakeup Call Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Hotel/Motel Room Remote Wakeup Call Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Individual Trunk Access:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Keep TelDir Entry on Check Out:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Local Music On Hold source:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Loudspeaker Pager Override:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Loudspeaker Pager Equivalent Zone Override Security:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Message Waiting:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Message Waiting Audible Tone Notification:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Message Waiting Deactivate On Off-Hook:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Message Waiting Inquire:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Multiline Set Loop Test:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Multiline Set Message Center Remote Read Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Multiline Set Music:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Multiline Set On-hook Dialing:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Multiline Set Phonebook Allowed:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Multiline Set Voice Mail Callback Message Erasure Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Name Suppression on outgoing Trunk Call:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Non DID Extension:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Non-Prime Public Network Identity:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Non Verified Account Code:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Off-Hook Voice Announce Allowed:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>ONS CLASS/CLIP: Message Waiting Activate/Deactivate:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>ONS CLASS/CLIP: Set:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>ONS CLASS/CLIP: Visual Call Waiting:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>ONS/OPS Internal Ring Cadence for External Callers:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes

<b>Override Interconnect Restriction on Transfer:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Pager Access All Zones:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Pager Access Individual Zones:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Privacy Released:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Public Network Access via DPNSS:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Public Network Identity Provided:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Public Network To Public Network Connection Allowed:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Public Trunk:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>R2 Call Progress Tone:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Record-A-Call Active:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Record-A-Call - Start Recording Automatically:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Record-A-Call - Save Recording on Hang-up:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Recorded Announcement Device:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Recorded Announcement Device - Advanced:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Redial Facilities:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Ringing Line Select:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>SC1000 Attendant Basic Function Key:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>SMDR External:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>SMDR Internal:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Speak@Ease Preferred:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Suite Services Enabled:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Suppress Simulated CCM after ISDN Progress:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Third Party Call Forward Follow Me Accept:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Third Party Call Forward Follow Me Allow:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Timed Reminder Allowed:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Trunk Calling Party Identification:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Trunk Flash Allowed:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Use Held Party Device for Call Re-routing:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>Use Called Party Call Hold Timer:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Voice Mail Softkey:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Account Code Length:</b>	<input type="text" value="12"/>	
<b>After Answer Display Time:</b>	<input type="text"/>	

Answer Plus Delay To Message Timer:	20
Answer Plus Expected Off-hook Timer:	30
Answer Plus Message Length Timer:	10
Answer Plus System Reroute Timer:	0
Attendant Busy Out Timer:	10
Auto Campon Timer:	10
Autovon Precedence:	4
Busy Tone Timer:	30
Call Duration:	10
Call Duration Forced Cleardown Timer:	0
Call Forward - Delay:	0
Call Forward No Answer Timer:	15
Call Hold Timer:	30
Campon Recall Timer:	10
Delay Ring Timer:	10
Dialing Conflict Timer:	3
Display Caller ID On Multicall/Keylines Timer:	5
Emergency Call - Audio Level for Set:	Ringer ▾
First Digit Timer:	15
Inter Digit Timer:	10
Lockout Timer:	45
ACD 2000 Logout Agent No Answer Timer:	15
Message Waiting Ringing Start Time Hour:	▾
Message Waiting Ringing Start Time Minute:	▾
Message Waiting Ringing Stop Time Hour:	▾
Message Waiting Ringing Stop Time Minute:	▾
No Answer Recall Timer:	17
ONS VMail-Delay Dial Tone Timer:	5
Ringing Timer:	180
Work Timer:	0
Key A:	
Key B:	
Key C:	
Key D:	

Save Cancel

Before moving on to the next task be sure to click the Save button.

## 6.2 Configuring ISDN-PRI Interface

Use the *ISDN PRI Link Descriptor Assignment* menu selection to build a template that sets the various options for a trunk interface such as framing, coding and interface type. Use the *Add* or *Change* buttons to either build a new Link Descriptor or modify an existing Link Descriptor.



IPBX3 - Mitel Networks® 3300 Integrated Communications Platform (ICP) - Microsoft Internet Explorer

Alarm Status: ✖ Critical 2006-Jul-27 10:04:13 Print... Export... Data Refresh Help Exit

IPBX3 - System Message:

Selection: System Configuration

- System Configuration
  - System Capacity
  - Units/Modules
    - Controller Module Configur...
    - Analog Services Unit Config...
    - Network Services Unit Conf...
    - Framer Configuration
    - Peripheral/DSU Unit
    - Controller Registry Configu...
  - Trunks
    - Class of Service Options As...
    - Analog Trunks
    - Digital Trunks
      - DPNSS
      - E1/CEPT
      - ISDN-BRI
      - ISDN-PRI
        - Link Descriptor Assig...
        - Digital Link Assignme...
        - MSDN-DPNSS-DASS
        - Trunk Service Assignm...
        - Digital Trunk Assignm...

Add Change Copy Delete

Previous Page 1 of 1 Next Go to: value: Go

Link Descriptor Assignment				
Number	Address for Message Control	BER - Maintenance Limit, 10 <sup>ms</sup> -n	BER - Service Limit, 10 <sup>ms</sup> -n	Data Call Alternate Digit Inversion
1	A	4	3	Yes
2	A	4	3	Yes
3	A	4	3	Yes
4	A	4	3	Yes

Link Descriptor Assignment

Number: 1  
 Address for Message Control: A  
 BER - Maintenance Limit, 10<sup>ms</sup>-n: 4  
 BER - Service Limit, 10<sup>ms</sup>-n: 3  
 Data Call Alternate Digit Inversion: Yes  
 Framing Losses in 24 hrs - Maintenance Limit: 255  
 Framing Losses in 24 hrs - Service Limit: 9000  
 Integrated Digital Access: ISDN NODE  
 Vendor Inter-working Type (Philips SOPHO): No  
 Satellite Link Delay: No  
 Signaling Maintenance Limit (line/24hr): 6000

MITEL 3300 ICP  
About System Administration

See the screen below for the selected options.

-- Web Page Dialog

### Link Descriptor Assignment

Number: 1

Address for Message Control: A

BER - Maintenance Limit, 10<sup>xx</sup>-n: 4

BER - Service Limit, 10<sup>xx</sup>-n: 3

Data Call Alternate Digit Inversion:  No  Yes

Framing Losses in 24 hrs - Maintenance Limit: 255

Framing Losses in 24 hrs - Service Limit: 9000

Integrated Digital Access: ISDN NODE

Vendor Inter-working Type (Philips SOPHO):  No  Yes

Satellite Link Delay:  No  Yes

Slip Rate - Maintenance Limit (slips/24hr.): 5000

Slip Rate - Service Limit (slips/24hr.): 7000

Alarm Debounce Timer - Service Limit (millisec.): 500

Voice Encoding: Invert

Data Encoding: Invert

QSIG Private Network Access:  No  Yes

Digital Link Fault Delay Timer (sec.): 240

Termination Mode:  LT  NT

Send Malicious Call Indication to PSTN for Tagged Calls:  No  Yes

T1 Only:

    B8ZS Zero Code Suppression:  No  Yes

    Operation Mode: CSU

    CSU Tx Line Build-Out (dB.): 0

    DSX-1 Line Length (Ft.): 0-133

    Extended Super Frame:  No  Yes

    Inverted D channel ( DPNSS only ):  No  Yes

E1 Only:

    CRC-4 Enabled:  No  Yes

    E1 Line Length (Ft.): 0-133

    E1 Impedance (Ohms):  75  120

Save Cancel

Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI Digital Link Assignment* menu selection to connect a physical location (port, unit, shelf, slot and link) within the PBX with the NSU and Link Descriptor configured in the previous steps. Select a *Controller Module* and click the *Change* button to configure the options.

Alarm Status: ✖ Critical 2006-Jul-27 10:01:13

Selection: System Configuration

System Configuration

- System Configuration
  - System Capacity
  - Units/Modules
    - Controller Module Configur...
    - Analog Services Unit Config...
    - Network Services Unit Conf...
    - Framer Configuration
    - Peripheral/DSU Unit
    - Controller Registry Configu...
  - Trunks
    - Class of Service Options As...
    - Analog Trunks
    - Digital Trunks
      - DPNSS
      - E1/CEPT
      - ISDN-BRI
      - ISDN-PRI
        - Link Descriptor Assig...
        - Digital Link Assignme...
        - MSDN-DPNSS-DASS
        - Trunk Service Assignm...
        - Digital Trunk Assignm...

Change Change Page Change All Clear

Previous Page 1 of 1 Next Go to: Change value: Go

Controller Module	Port	Unit	Shelf	Slot	Link	Interface Type	Digital Link Descriptor	Comment
1	1	6	1	1	1	UNIVERSAL T1	1	T1 QSIG
1	1	6	1	1	2	UNIVERSAL T1	3	T1 NI2
1	2	7	1	1	1	UNIVERSAL E1	2	E1 QSIG
1	2	7	1	1	2	UNIVERSAL E1	4	DK &RS

MITEL 3300 ICP  
About System Administration

See the screen below for the selected options.

-- Web Page Dialog

**Digital Link Assignment**

<b>Controller Module:</b>	1
<b>Port:</b>	1
<b>Unit:</b>	6
<b>Shelf:</b>	1
<b>Slot:</b>	1
<b>Link:</b>	1
<b>Interface Type:</b>	UNIVERSAL T1
<b>Digital Link Descriptor:</b>	<input type="text" value="1"/>
<b>Comment:</b>	<input type="text" value="T1 QSIG"/>

Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI MSDN-DPNSS-DASSII Trunk Circuit Descriptor* menu selection to assign direction and protocols to the individual trunk cards in the PBX. Select a trunk number and click the Change button to configure the options.

The screenshot shows the Mitel 3300 ICP web interface. At the top, there is an alarm status: **Critical 2006-Jul-27 10:01:13**. Below this is the **Selection:** dropdown menu set to **System Configuration**. The left-hand navigation menu includes options like **Configuration**, **Capacity**, **Modules**, **Controller Module Configuration**, **Log Services Unit Configuration**, **Work Services Unit Configuration**, **Printer Configuration**, **Peripheral/DSU Unit**, **Controller Registry Configuration**, **Assignment of Service Options**, **Assignment of Trunks**, **Digital Trunks**, **ISDN-DPNSS**, **ISDN-CEPT**, **ISDN-BRI**, **ISDN-PRI**, **Link Descriptor Assignment**, **Digital Link Assignment**, **MSDN-DPNSS-DASSII Trunk Circuit Descriptor** (which is highlighted), **Trunk Service Assignment**, and **Digital Trunk Assignment**.

The main content area displays a table titled **MSDN-DPNSS-DASSII Trunk Circuit Descriptor**. The table has the following columns: **Number**, **Card Type**, **Dual Seizure Priority**, **Far End Connection**, **Signalling Protocol**, and **ISDN BRI Mode**. The table contains three rows of data:

Number	Card Type	Dual Seizure Priority	Far End Connection	Signalling Protocol	ISDN BRI Mode
1	UNIVERSAL T1	Incoming	Local Office	MSDN-DPNSS	
2	UNIVERSAL E1	Incoming	Local Office	MSDN-DPNSS	
3	UNIVERSAL E1	Incoming	Local Office	MSDN-DPNSS	

At the bottom of the interface, the Mitel logo and '3300 ICP About System Administration' are visible.

See the screen below for the selected options.

-- Web Page Dialog

**MSDN-DPNSS-DASSII Trunk Circuit Descriptor**

**Number:** 1

**Card Type:** UNIVERSAL T1

**Dual Seizure Priority:**  Incoming  Outgoing

**Far End Connection:** Local Office

**Signalling Protocol:**  MSDN-DPNSS  DASS II

**ISDN BRI Mode:**

Save Cancel

Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI Trunk Service Assignment* menu selection to set up a template that contains various service levels, for example the trunks class of service that was previously defined, into a template that will latter on be assigned to a trunk. Select the trunk service number and click the Change button to configure the options.

The screenshot shows the IPBX3 System Message configuration interface. The left sidebar contains a tree view of system configuration options, with 'ISDN-PRI' expanded to show 'Trunk Service Assignment'. The main content area displays a table of trunk service assignments and a configuration summary for the selected entry.

Trunk Service Number	Release Link Trunk	Class of Service	Class of Restriction	Baud Rate	Intercept Number	Trunk Label
1	No	1	1	300	1	Radbrook
2	No	1	1	300	1	3 to 4
3	No	1	1	300	1	3 to 5
4	No	1	1	300	1	ASU Lp Bk
5	No	5	1	300	1	E1 QSIG
6	No	6	1	300	1	T1 NI2
7	No	1	1	300	1	
8	No	1	1	300	1	
9	No	1	1	300	1	
10	No	5	1	300	1	DK &RS

**Trunk Service Assignment**

Trunk Service Number: 1  
 Release Link Trunk: No  
 Class of Service: 1  
 Class of Restriction: 1  
 Baud Rate: 300  
 Intercept Number: 1  
 Non-dial In Trunks Answer Point - Day:  
 Non-dial In Trunks Answer Point - Night 1:  
 Non-dial In Trunks Answer Point - Night 2:  
 Dial In Trunks Incoming Digit Modification - Absorb: 0  
 Dial In Trunks Incoming Digit Modification - Insert:

See the screen below for the selected options.

-- Web Page Dialog

### Trunk Service Assignment

Trunk Service Number:	1
Release Link Trunk:	No
Class of Service:	1
Class of Restriction:	1
Baud Rate:	300
Intercept Number:	1
Non-dial In Trunks Answer Point - Day:	
Non-dial In Trunks Answer Point - Night 1:	
Non-dial In Trunks Answer Point - Night 2:	
Dial In Trunks Incoming Digit Modification - Absorb:	0
Dial In Trunks Incoming Digit Modification - Insert:	
Trunk Label:	T1 QSIG

Save Cancel

Before moving on to the next task be sure to click the Save button.



Use the *ISDN-PRI Digital Trunk Assignment* menu selection to configure the individual trunk members and assign them the defined *Trunk Service Assignment* template and *Trunk Circuit Descriptor* template that were configured in the previous steps. This gets done for each member of a trunk interface. Select each individual trunk member and click the Change button to configure the options.

The screenshot shows the IPBX3 System Administration interface. The left-hand navigation tree is expanded to 'ISDN-PRI' > 'Digital Trunk Assignment'. The main content area displays a table of trunk assignments and a detailed configuration view for a selected member.

Cabinet	Shelf	Slot	Circuit	Card Type	Trunk Number
6	1	1	1	UNIVERSAL T1	100
6	1	1	2	UNIVERSAL T1	101
6	1	1	3	UNIVERSAL T1	102
6	1	1	4	UNIVERSAL T1	103
6	1	1	5	UNIVERSAL T1	104
6	1	1	6	UNIVERSAL T1	105
6	1	1	7	UNIVERSAL T1	106
6	1	1	8	UNIVERSAL T1	107
6	1	1	9	UNIVERSAL T1	108
6	1	1	10	UNIVERSAL T1	109

<b>Digital Trunk Assignment</b>	
Cabinet:	6
Shelf:	1
Slot:	1
Circuit:	1
Card Type:	UNIVERSAL T1
Trunk Number:	100
Trunk Service Number:	1
DTS Service Number:	
Circuit Descriptor Number:	1
Interconnect Number:	1

See the screen below for the selected options.

**Range Programming -- Web Page Dialog**

**Change Range Programming - Digital Trunk Assignment**

This form allows you to change one or more records, starting at the following record:

Cabinet	Shelf	Slot	Circuit	Card Type	Trunk Number	Trunk Service Number	DTS Service Number
6	1	1	1	UNIVERSAL T1	100	1	

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
<b>Cabinet:</b>	-	6	-
<b>Shelf:</b>	-	1	-
<b>Slot:</b>	-	1	-
<b>Circuit:</b>	-	1	-
<b>Card Type:</b>	-	UNIVERSAL T1	-
<b>Trunk Number:</b>	Change to <input type="text" value="v"/>	<input type="text" value="100"/>	<input type="text" value=""/>
<b>Trunk Service Number:</b>	Change to <input type="text" value="v"/>	<input type="text" value="1"/>	<input type="text" value=""/>
<b>DTS Service Number:</b>	Change to <input type="text" value="v"/>	<input type="text" value=""/>	<input type="text" value=""/>
<b>Circuit Descriptor Number:</b>	Change to <input type="text" value="v"/>	<input type="text" value="1"/>	<input type="text" value=""/>
<b>Interconnect Number:</b>	Change to <input type="text" value="v"/>	<input type="text" value="1"/>	<input type="text" value=""/>

Before moving on to the next task be sure to click the Save button.

## 6.3 Configuring ARS Options

Use the *ARS Trunk Group Assignment* menu selection to edit your trunk group configuration. Use the Add or Change buttons to either build a new trunk group or modify an existing group.

The screenshot displays the IPBX3 - System Administration web interface. The left sidebar shows a tree view with 'Automatic Route Selection (ARS)' expanded, and 'Trunk Group Assignment' selected. The main content area shows a table of Trunk Group Assignments with columns for Trunk Group Number, Hunt Mode, Trunk Group Busy RAD, Maximum Network Hop, and Comments. Below the table are navigation buttons (Previous, Next), a 'Go to' search field, and buttons for 'Add Member', 'Change Member', and 'Delete Member'. A second table, 'Trunk Group Members', lists members 1 through 10 with their corresponding Trunk Numbers (100-109).

Trunk Group Number	Hunt Mode	Trunk Group Busy RAD	Maximum Network Hop	Comments
15	Terminal			
21	Terminal			
22	Terminal			
23	Terminal			
24	Terminal			

Member	Trunk Number
1	100
2	101
3	102
4	103
5	104
6	105
7	106
8	107
9	108
10	109

See the screen below for the selected options.

**Range Programming -- Web Page Dialog** ✖

**Change Range Programming - Trunk Group Assignment** Help

This form allows you to change one or more records, starting at the following record:

Trunk Group Number	Hunt Mode	Trunk Group Busy RAD	Maximum Network Hop	Comments
21	Terminal			

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
<b>Trunk Group Number:</b>	Change to ▼	<input type="text" value="21"/>	<input type="text"/>
<b>Hunt Mode:</b>	Change to ▼	<input checked="" type="radio"/> Terminal <input type="radio"/> Circular	-
<b>Trunk Group Busy RAD:</b>	Change to ▼	<input type="text"/>	<input type="text"/>
<b>Maximum Network Hop:</b>	Change to ▼	<input type="text"/>	<input type="text"/>
<b>Comments:</b>	Change to ▼	<input type="text"/>	-

Preview Save Cancel

Click the Save button when you are finished editing the options and wish to save your configuration.

When you have configured a trunk group you use the Add Member button to add individual trunk members to the group.

Range Programming -- Web Page Dialog

Change Range Programming - *Trunk Group Members* Help

This form allows you to change one or more records, starting at the following record:

Trunk Number  
100

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Trunk Number:	Change to	100	

Preview Save Cancel

Before moving on to the next task be sure to click the Save button.

Use the *ARS Route Assignment* menu selection to define a route to direct calls to a specific trunk. Select a Route Number and click the Change button to configure the options.

Alarm Status: ✖ Critical 2006-Jul-27 10:01:13

Selection: System Administration

System Administration

- System Administration
  - System Audio Files Update
  - User Authorization Profiles
  - System Options
    - Automatic Route Selection (ARS)
      - Trunk Service Assignment
      - Trunk Group Assignment
      - Class of Restriction Group
      - System Account Code Defir
      - Independent Account Code
      - Default Account Code Defini
      - Call Progress Tone Detecti
      - Digit Modification Assignme
      - Maximum Dialed Digits
      - Route Assignment
      - Route List Assignment (Op
      - Day and Time Zone Assign
      - Route Plan Assignment (Op
      - ARS Digits Dialed Assignm
      - ARS Leading Digits Assign
      - Interconnect Restriction
      - Node Identity Assignment

Change Change Page Change All Clear

Previous Page 1 of 8 Next Go to: value: Go

Route Number	Trunk Group Number	COR Group Number	Digit Modification Number	Digits Before Outpulsing	XNET Trunk Group Number	Route Type	Compression
1		1	2		1		Off
2		1	2		2		Off
3		1	2		3		Off
4	21	1	1	5			Off
5	22	1	1	5			Off
6	23	1	1	5			Off
7		1	1				Off
8		1	1				Off
9		1	1				Off
10	24	1	1	5			Off
11		1	1				Off
12		1	1				Off
13		1	1				Off
14		1	1				Off
15	15	1	1				Off

MITEL 3300 ICP  
About System Administration

See the screen below for the selected options.

-- Web Page Dialog

### Route Assignment

Route Number: 4

Trunk Group Number: 21

COR Group Number: 1

Digit Modification Number: 1

Digits Before Outpulsing: 5

XNET Trunk Group Number:

Route Type:

Compression:  Off  On

Save Cancel

Before moving on to the next task be sure to click the Save button.

Use the *ARS Digits Dialed Assignment* menu selection to configure an ARS number to use to place and forward calls to a specific trunk group. The ARS number is used as the forwarding target for subscriber station sets and the inbound entry point for direct calls to the server. Use the Add or Change buttons to either add a new ARS number or modify an existing one.

The screenshot shows the Mitel Networks 3300 ICP web interface. The main content area is titled "ARS Digits Dialed Assignment" and contains a table with the following data:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
4	2	Route	15
5	2	Route	10
6	2	Route	5
7	2	Route	22
8	2	Route	21
92	4	Route	1
94	4	Route	2
95	4	Route	3

The interface also shows a left-hand navigation tree with "ARS Digits Dialed Assignment" selected. At the top right, there are buttons for "Add", "Change", and "Delete". The page is displayed in Microsoft Internet Explorer.

This example shows setting up ARS to except any 3 digit number that starts with an 8 as a dialable number. The ASR table will then take the call and route it to trunk group 21. Useable numbers in this example would be any number between 800 and 899 all inclusive.

An alternate method of configuration would be to define a very specific number, for example 800, not an entire range, and not define any following digits.

The method you choose is up to what your sites configuration will support.

See the screen below for the selected options.



**Range Programming -- Web Page Dialog** ✖

**Change Range Programming - ARS Digits Dialed Assignment** Help

This form allows you to change one or more records, starting at the following record:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
8	2	Route	21

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
<b>Digits Dialed:</b>	Change to ▼	<input type="text" value="8"/>	<input type="text" value=""/>
<b>Number of Digits to Follow:</b>	Change to ▼	<input type="text" value="2"/> ▼	-
<b>Termination Type:</b>	Change to ▼	Route ▼	-
<b>Termination Number:</b>	Change to ▼	<input type="text" value="21"/>	<input type="text" value=""/>

Preview Save Cancel

Below is an example of the configuration using the described alternative method.

**Range Programming -- Web Page Dialog** ✖

**Change Range Programming - ARS Digits Dialed Assignment** Help

This form allows you to change one or more records, starting at the following record:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
800		Route	21

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
<b>Digits Dialed:</b>	Change to ▼	<input type="text" value="800"/>	<input type="text"/>
<b>Number of Digits to Follow:</b>	Change to ▼	<input type="text" value=""/>	-
<b>Termination Type:</b>	Change to ▼	Route ▼	-
<b>Termination Number:</b>	Change to ▼	<input type="text" value="21"/>	<input type="text"/>

Preview Save Cancel

Before moving on to the next task be sure to click the Save button.

## 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 the phone's soft menu keys. The subscriber should be directed to set their internal and external ring no answer and busy forwarding conditions to the Pilot Number setting defined in the hunt group configuration.

## 6.5 Additional Comments

Ensure that the Node Identity assignment has not been entered. If this is entered, it will append extra digits onto extensions as they pass across the trunk to the gateway.

## 7. Microsoft OCS setup

### 7.1 Steps for configuring OCS

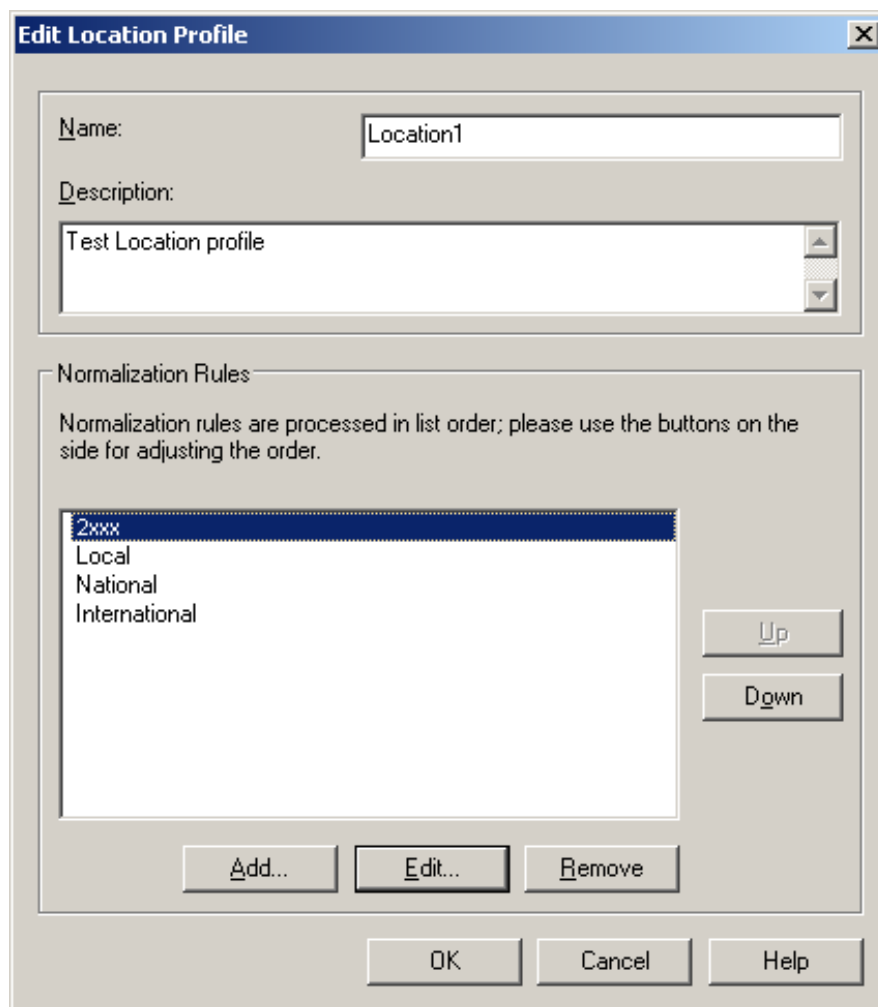
Normalization rules are used to convert all possible dial numbers into full E.164 formatted numbers. Microsoft OCS uses the standard E.164 format to search for all users listed in Active Directory (AD).

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` from the menu provided. Edit a location profile as shown in the example below.



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

**Edit Phone Number Normalization Rule**

Name:

Click to copy an existing rule.

Description:

Translation

Phone pattern regular expression:

Translation pattern regular expression:

Valid translation characters are +, numbers, and \$. Example: +1425\$1.

Click Helper for assistance in creating common phone number regular expressions and translations.

Test translation

To test the translation, enter a sample dialed number. If it matches the phone pattern, the translation will be shown.

Sample dialed number:

Translated number:

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	<code>^(2[0-9]{3})\$</code>	<code>+1716639\$1</code>	Normalize 2xxx to E.164
Local	<code>^(\\d{7})\$</code>	<code>+1716\$1</code>	Local number
National	<code>^1(\\d*)\$</code>	<code>+1\$1</code>	Long distance number
International	<code>^011(\\d*)</code>	<code>+011\$1</code>	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* from the menu provided. Edit a route as shown in the example below.

**Edit Route**

Name:

Description:

A route requires a target phone number regular expression, one or more gateways, and one or more phone usages.

Target phone numbers:

Target regular expression:

Helper...

Gateways

Address
dmg4000.BufOCS.local:5061

Add... Remove

Phone usages

Default Usage

Configure...

OK Cancel Help

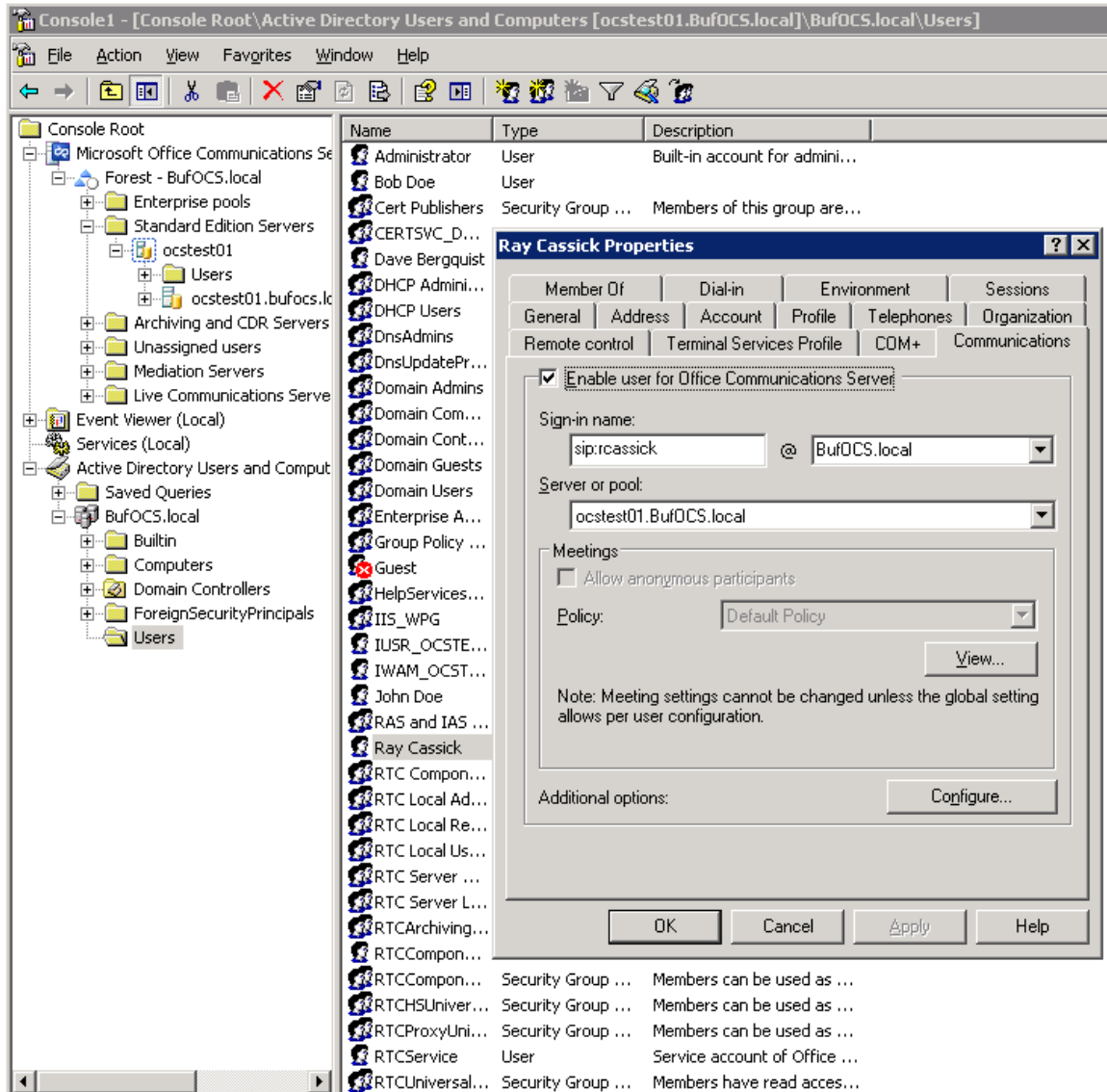
This entry routes any number with or without '+' prefix followed by any digits to Mediation server dmg4000.bufocs.local

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.

**User Options** [X]

**Telephony**  
 Select a telephony option. These settings affect only those calls that are routed through IP-PSTN or remote call control gateways.

Enable PC-to-PC communication only  
 Enable Remote call control  
 Enable Enterprise Voice  
 Enable PBX integration

Note: To enable both remote call control and PBX integration, you must specify a Server URI below.

Policy:

Server URI:

Line URI:

**Federation**

Enable federation  
 Enable remote user access  
 Enable public IM connectivity

**Archiving**

Archive internal IM conversations  
 Archive federated IM conversations

Note: Archiving settings cannot be changed unless the global setting allows per user configuration.

Enable enhanced presence

Note: Enhanced presence cannot be changed once it has been set.

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
<b>Inbound call scenarios</b>		
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:
  - 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:



- 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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