Interactive Intelligence Interaction Center 2.4

PBX Configuration Note:

Nortel Communication Server 1000M with IC using SIP

Technical Reference

By Interactive Intelligence, Inc.

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Customer Interaction Center®
Vonexus Enterprise Interaction Center®

Document Version 1.1

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Content

This document describes the configuration required to setup Nortel CS-1000 to interoperate with Interactive Intelligence Interaction Center 2.4 (IC) using SIP.
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Interaction Center Platform Statement

This document describes Interaction Center (IC) features that may not be available in your IC product. Several products are based on the IC platform, and some features are disabled in some products.

These products are based on the IC platform:

- Customer Interaction Center (CIC)
- Vonexus Enterprise Interaction Center (Vonexus EIC, or EIC)
- Messaging Interaction Center (MIC)

While all of these products share a common feature set, this document is intended for use with all IC products, and some of the described features may not be available in your product.
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Who should read this document

This document is intended for Systems Integrators with significant telephony knowledge.

Technical Support

The information contained within this document has been provided by Interactive Intelligence, its partners or equipment manufacturers and is provided AS IS. This document contains information about how to modify the configuration of your PBX. Improper configuration may result in the loss of service of the PBX. Interactive Intelligence is unable to provide support or assistance with the configuration or troubleshooting of components described within. Interactive Intelligence recommends readers to engage the service of an Interactive Intelligence IC Certified Engineer or the manufacturers of the equipment(s) described within to assist with the planning and deployment of Messaging Interaction Center.

Known Issues

Nortel may include user attribute information in its SIP headers, which prevents successful integration on inbound calls to xIC. TS should be removing these attributes before passing the information to Interaction Processor, where handlers have access to it. The following SCRs correct this behavior if you encounter it:

SCR 65714 (2.4)

SCR 65715 (3.0)

For part of the tests, we attempted to use the Nortel I2050 PC Softphone (Version 2.01.0255). IC could not recognize DTMF from this softphone. No other devices presented this issue.
Chapter 1: General Information

Components

PBX or IP-PBX

<table>
<thead>
<tr>
<th>PBX Vendor</th>
<th>Nortel</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model</td>
<td>Signaling Server Version 3621</td>
</tr>
<tr>
<td></td>
<td>Communication Server 1000M MG/CP PIV CP PIV - Pentium M 1.1 GHz</td>
</tr>
<tr>
<td>Software Version</td>
<td>RELEASE 5 ISSU 50 J + IDLE_SET_DISPLAY</td>
</tr>
<tr>
<td>Telephony Signaling</td>
<td>SIP</td>
</tr>
<tr>
<td>Additional Notes</td>
<td>None</td>
</tr>
</tbody>
</table>

Interactive Intelligence Interaction Center

| Version | 2.4 SU 33 |

Prerequisites

PBX Prerequisites

- SIP option

Summary and Limitations

Nortel may include user attribute information in its SIP headers, which prevents successful integration on inbound calls to xIC. TS should be removing these attributes before passing the information to Interaction Processor, where handlers have access to it. The following SCRs correct this behavior if you encounter it:

SCR 65714 (2.4)

SCR 65715 (3.0)
Chapter 2: xIC Setup

Step 1: Create the SIP Line

Make sure to note the Domain Name used on the SIP Line. It will be needed in the Nortel configuration.
Make sure to check the "Use Numeric User Portion for Telephone Number checkbox.

The transport can be TCP or UDP to communicate with the Nortel – only TCP was tested.

The line does NOT need to register with the Nortel.
To configure the Route List Block (RLB):

In the menu frame, click on **Dialing and Number Plans** -> **Electronic Switched Network**.

In the main display frame, click on **Route List Block (RLB)**.
Add a new RLB by providing a new route book list index and click on **to Add**.

'14' is used as the route list block index in this example.

Click on **Edit** beside the new RLB entry created.
Configure the properties of the RLB with following values:

Entry Number for the Route List (ENTR) 0

Route Number (ROUT) 14

Click on Submit.
To configure the steering code:

In the menu frame, click on **Dialing and Number Plans -> Electronic Switched Network**.

In the main display frame, click on **Distance Steering Code (DSC)**.

Select option **Add**.

Provide the initial digit of the UM pilot or/and Auto-Attendant extensions.

In this example, Subscriber Access DSC ‘8500’.

Click on **to Add**.

Do the same for Auto Attendant DSC ‘8501’

Click on **to Add**.
Configure the properties of Distant Steering Code with the following values:

**Flexible Length Number of Digits (FLEN)**  4 (*corresponds to the length of 8500*)

**Route List to be Accessed (RLI)**  14 (*corresponds to the RLB created*)

Click on **Submit**.
In the menu frame, click on **System -> IP Network -> Nodes: Servers, Media Cards**.

In the main display frame, look for the node entry which corresponds to the signaling server that will contact IC. This is the same signaling server which is configured as the Proxy for the SIP line in Interaction Administrator that will communicate with this Nortel.

Click on **Edit...** which corresponds to this node entry.
Click to expand the **Microsoft Unified Messaging** tab as illustrated in the diagram above.

Configure the properties under the tab with the following values:

- **Subscriber Access Number** 8500
- **Auto Attendant Number** 8501

If MWI application is going to be used for Lamp Status Update, configure the properties:

- **MWI Application DN** 8500
- **MWI Dialing Plan** CDP
Make note of the **SIP Domain name** it will need to be the same in the NRS server and on the SIP line in Interaction Administrator (case sensitive).

**Click on Save and Transfer.**
Logon to the NRS.

Click on **Configuration > Gateway Endpoints >Show**.

In the main display frame, ensure the **Standby database** radio button is selected.

Under Gateway Endpoints, click on **Add...** to create an entry for **Messaging**.
Configure the properties of the endpoint with the following values:

- **Endpoint Name**: IC or ININ Proxy Server (Unique Name)
- **Endpoint authentication enabled**: Authentication Off
- **Static endpoint address**: <IP Address of the Exchange Server>
SIP support  
**Static SIP endpoint**

SIP TCP transport enabled  
*Checked*

SIP TCP port  
5060

Click on **Save**.

Go to **Routes**.

Click on **Add**... to create a new routing entry. The routing entry informs the SPS/SRS to route all calls for the Messaging Pilot Number and Auto Attendant Number to Messaging Server.
Configure the following properties as illustrated in the diagram above:

DN type  
Private level 0 Regional (CDP steering code)

DN prefix  
850

Route cost  
1

Click on Save.
Go to **Configuration > Tools > Database Actions.**

Click on **Cut Over**, followed by **Commit**, to update the changes to the database.

**Message Center Programming**
Create a Message center ACD Queue and NCFW to your Subscriber Access DSC. Example used here is: ‘8500’

This number will be your **Subscriber Access Number** and will be the number you use when programming the MWK key on your phone sets.

```
REQ   prt
TYPE  acd
CUST  0
ACDN  8200
TYPE  ACD
CUST  0
ACDN  8200
MWC   YES
MAXP  1
SDNB  NO
BSCW  NO
ISAP  NO
AACQ  NO
RGAI  NO
ACAA  NO
```
FRRT
SRRT
NRRT
FROA NO
CALP POS
ICDD NO
NCFW 8500
FNCF NO
FORC NO
RTQT 0
SPCP NO
OBTN NO
RAO NO
CWT 1
NCWL NO
BYTH 0
OVTH 2047
TOFT NONE
HPQ NO
OCN NO
OVDN
IFDN
OVBU LNK LNK LNK LNK
EMRT
MURT
RTPC NO
STIO
TSFT 20
HOML YES
RDNA NO
LABEL KEY NO
NRAC NO
DAL NO
RPRT NO
RAGT 4
DURT 30
RSND 4
FCTH 20
CRQS 100
IVR NO
OBSC NO
OBPT 5
CWNT NONE
Auto Attendant Programming

Create a Message center ACD Queue and NCFW to your Auto Attendant Access DSC.
Example used here is: ‘8501’

This number will be your Auto Attendant Access number.

REQ  prt
TYPE  acd
CUST  0
ACDN  8201

TYPE  ACD
CUST  0
ACDN  8201
MWC   YES
MAXP  1
SDNB  NO
BSCW  NO
ISAP  NO
AACQ  NO
RGAI  NO
ACAA  NO
FRRT  
SRRT  
NRRT  
FROA  NO
CALP  POS
ICDD  NO
NCFW  8501
FNCF  NO
FORC  NO
RTQT  0
SPCP  NO
OBTN  NO
RAO   NO
CWTH  1
NCWL  NO
BYTH  0
OVTH  2047
TOFT  NONE
HPQ   NO
OCN   NO
OVDN  
IFDN  
OVBU  LNK LNK LNK LNK
EMRT  
MURT  
RTPC  NO
STIO  
TSFT 20
HOML YES
RDNA NO
LABEL_KEY0 NO
NRAC NO
DAL NO
RPRT NO
RAGT 4
DURT 30
RSND 4
FCTH 20
CRQS 100
IVR NO
OBSC NO
OBPT 5
CWNT NONE

Station Programming Example

REQ: prt
TYPE: tn
TYPE TNB
TN 840121
DATE
PAGE
DES
DES TEST
TN 08401201
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
ERL 0
FDN 8200
TGAR 1
LDN NO
NCOS 2
SGRP 0
RNPG 4
SCI 0
SSU
LNRS 16
XLST
CLS UNR FBD WTA LPR PUA MTA FNA HTA TDD HFA GRLD CRPD STSD MWA LMPN RMMD SMWD AAD IMD XHD IRA NID OLA VCE DRG1 POD DSX VMD SLKD CCSD SWD LNA CNDA CFTD SFA MRD DDV CNNID CDCA MSID DAPA BFED RCBD ICDD CDMD LLCN MCTA CLBD AUTU GPUD DPUD DNDD CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
DDGA NAMA
DRDD EXR0
USMD USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
CDMR MCDD T87D PKCH
CPND_LANG ENG
RCO 0
HUNT 8200
LHK 1
LPK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 4116 0 MARP
CPND
NAME Test Phone
XPLN 18
DISPLAY_FMT FIRST,LAST
01 SCR 36116 1 MARP
CPND
NAME Test Phone
XPLN 18
DISPLAY_FMT FIRST,LAST
02 ADL 16
03 MIK
04 MCK
05 TRC
06 ADL 16
07 ADL 16
08
09
10
11
12
13
14
15
16 MWK 8200
17 TRN
18 AO6
19 CFW 8 8200
20 RGA
21 PRK
22 RNP
23 SCC 0999
24 PRS
25 CHG
26 CPN
27 CLT
28 RLT
29
30
31
Chapter 4: Messaging Interaction Center 2.4 Validation Test Matrix

Testing the Core Feature Set

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (P)
- Conditional Pass (CP)
- Fail (F)
- Not Tested (NT)
- Not Applicable (NA)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.
- Section 6.1 for detailed descriptions of call scenario failures, if any.

<table>
<thead>
<tr>
<th>No.</th>
<th>Call Scenarios (see appendix for more detailed instructions)</th>
<th>(P/CP/F/NT)</th>
<th>Reason for Failure (see 6.1 for more detailed descriptions)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user’s mailbox. Confirm hearing the prompt: “Welcome to Communité. To access your mailbox, enter your extension...”</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Navigate mailbox using Mobile Office</td>
<td>NT</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Navigate mailbox using the Telephony User Interface (TUI).</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>Dial user extension and leave a voicemail.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>No.</td>
<td>Call Scenarios (see appendix for more detailed instructions)</td>
<td>(P/CP/F/NT)</td>
<td>Reason for Failure (see 6.1 for more detailed descriptions)</td>
</tr>
<tr>
<td>-----</td>
<td>------------------------------------------------------------</td>
<td>-------------</td>
<td>----------------------------------------------------------</td>
</tr>
<tr>
<td>4a</td>
<td>Dial user extension and leave a voicemail from an internal extension.</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>4b</td>
<td>Dial user extension and leave a voicemail from an external phone.</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>Dial Auto Attendant (AA).</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Dial the extension for the AA and confirm the AA answers the call.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>Call Transfer by Dial By Name.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6a</td>
<td>Call Transfer by Dial By Name and have the called party answer.</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Confirm the correct called party answers the phone.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6b</td>
<td>Call Transfer by Dial By Name when the called party’s phone is busy.</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Confirm the call is routed to the called party’s voicemail.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6c</td>
<td>Call Transfer by Dial by Name when the called party does not answer.</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Confirm the call is routed to the called party’s voicemail.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>No.</td>
<td>Call Scenarios (see appendix for more detailed instructions)</td>
<td>(P/CP/F/NT)</td>
<td>Reason for Failure (see 6.1 for more detailed descriptions)</td>
</tr>
<tr>
<td>-----</td>
<td>-------------------------------------------------------------</td>
<td>-------------</td>
<td>------------------------------------------------------------</td>
</tr>
<tr>
<td>7</td>
<td>Configure a button on the phone of a UM-enabled user to forward the user to the pilot number. Press the voicemail button._CONFIRM you are sent to the prompt: “Welcome to Communité. Please enter your passcode.”</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>Send a test Fax message to user extension.</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Confirm the Fax is received in the user’s inbox.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>Setup Message Waiting Indicator (MWI).</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>Blind Transfer</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>11</td>
<td>Consult Transfer</td>
<td>P</td>
<td></td>
</tr>
<tr>
<td>12</td>
<td>Dynamic Audio</td>
<td>P</td>
<td></td>
</tr>
</tbody>
</table>

**Detailed Description of Limitations**

<table>
<thead>
<tr>
<th>Failure Point</th>
<th>None</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phone type (if phone-specific)</td>
<td></td>
</tr>
<tr>
<td>Call scenarios(s) associated with failure</td>
<td></td>
</tr>
</tbody>
</table>
Appendix
The test cases in this section were run against a Messaging Interaction Server. It should be noted, however, that the cases selected were specifically chosen because they exercise SIP messaging in such a way that satisfies requirements for any IC platform to integrate as a “SIP Tie Line”.

Dial Pilot Number and Mailbox Login
1. Dial the pilot number of the MIC server from an extension that is NOT enabled for Voicemail.
2. Confirm hearing the greeting prompt: “Welcome to Communité. Please enter your extension...”
3. Enter the extension, followed by the pound sign, and then the passcode of a Voicemail enabled user.
4. Confirm successful logon to the user’s mailbox.

Navigate Mailbox using Mobile Office
1. Logon to a user’s mailbox who is licensed for Mobile Office
2. Navigate through the mailbox and try out various voice commands to confirm that the Mobile Office is working properly.
3. This test confirms that the RTP is flowing in both directions and speech recognition is working properly.

Navigate Mailbox using Telephone User Interface (TUI)
1. Logon to a user’s mailbox.
2. Navigate through the mailbox and try out the various key commands to confirm that the TUI is working properly.
3. This test confirms that both the voice RTP and DTMF RTP (RFC 2833) are flowing in both directions.
Dial User Extension and Leave Voicemail

**Note:** If you are having difficulty reaching the user’s voicemail, verify that the coverage path for the user’s phone is set to the pilot number of the MIC server.

**From an Internal Extension**
1. From an internal extension, dial the extension for a Voicemail enabled user and leave a voicemail message.
2. Confirm the voicemail message arrives in the called user’s inbox.
3. Confirm this message displays a valid MIC user’s name as the sender of this voicemail.

**From an External Phone**
1. From an external phone, dial the extension for a Voicemail enabled user and leave a voicemail message.
2. Confirm the voicemail message arrives in the called user’s inbox.
3. Confirm this message displays the phone number as the sender of this voicemail.

**Dial Auto Attendant (AA)**
1. Create an Auto Attendant using the MIC Web Administrator:
2. Dial the extension of Auto Attendant.
3. Confirm the AA answers the call.

**Call Transfer by Dial By Name**
1. Dial the pilot number for the MIC server from a phone that is NOT associated with a MIC user.
2. To search for a user by name:
   - Press 2 to Dial By Name.
   - Call Transfer by Dial By Name by entering the name of an MIC user using the telephone keypad, last name first.

   **Note:** Even though some keys are associated with three or four numbers, for each letter, each key only needs to be pressed once regardless of the letter you want. Ignore spaces and symbols when spelling the name.

**Called Party Answers**
- Call Transfer by Dial By Name to a user in the same dial plan and have the called party answer.
3. Confirm the call is transferred successfully.

**Called Party is Busy**
1. Call Transfer by Dial By Name to a user in the same dial plan when the called party is busy.
2. Confirm the calling user is routed to the correct voicemail.

**Called Party does not Answer**
1. Call Transfer by Dial By Name to a user in the same dial plan and have the called party not answer the call.
2. Confirm the calling user is routed to the correct voicemail.

**Voicemail Button**
1. Configure a button on the phone of a Voicemail enabled user to route the user to the pilot number of the MIC server.
2. Press the voicemail button.
3. Confirm you are sent to the prompt: "Welcome to Communité. Please enter your passcode...”

   **Note:** If you are not hearing this prompt, verify that the button configured on the phone passes the user’s extension as the redirect number. This means that the user extension should appear in the diversion information of the SIP invite.

**Testing Fax Features**
To test fax functionality:
1. Dial the extension for a fax-enabled MIC user from a fax machine.
2. Confirm the fax message is received in the user’s inbox.

   **Note:** You may notice that the MIC server answers the call as though it is a voice call (i.e. you will hear: "Please leave a message for...”). When the MIC server detects the fax CNG tones, it switches into fax receiving mode, and the voice prompts terminate.

   **Note:** MIC only supports T.38 for sending fax.

**Message Waiting Indicator (MWI)**
1. Enable MWI for a Voicemail enabled user.
2. Leave a message for that user.
3. Verify MWI goes on
4. Delete or Mark Saved that message
5. Verify MWI goes off

   **Note:** MWI doesn’t go off until there are no more New messages in the Inbox.

**Blind Transfer**
1. Verify Putback is enabled on SIP Line.
2. Ring No Answer to a user’s voicemail.
3. Zero Out to user’s Operator.
5. Verify that all resources are released from IC.

**Consult Transfer**

1. Verify Putback is enabled on SIP Line, and that Follow Me is enabled for your test user.
2. Ring No Answer to a user’s voicemail.
3. Press 2 to Follow Me.

   **Note:** Follow me should be setup to an internal extension.

4. Answer Follow Me call.
5. Verify that all resources are released from IC.

**Dynamic Audio**

1. Verify Putback is **NOT** enabled on SIP Line, Dynamic Audio is enabled on the SIP Line, and that Follow Me is enabled for your test user.
2. Ring No Answer to a user’s voicemail.
3. Press 2 to Follow Me.

   **Note:** Follow me should be setup to an internal extension.

4. Answer Follow Me call.
5. Both legs of the call should be visible in Supervisor.
6. No IP resources should be in use.
# Appendix A: Change Log

<table>
<thead>
<tr>
<th>Change Log Date</th>
<th>Changes Made</th>
</tr>
</thead>
<tbody>
<tr>
<td>02-13-2009</td>
<td>Created document.</td>
</tr>
<tr>
<td>02-16-2009</td>
<td>Added updated PBX configuration screen shots.</td>
</tr>
</tbody>
</table>
Appendix B: Acronyms Used in This Document

Here are some of the most important acronyms used in this document.

- CAS: Centralized Attendant Service
- CNG: CalliNG tone sent by a fax machine
- DTMF: Dual Tone Multi-Frequency
- IA: Interaction Administrator
- IC: Interaction Center
- IP: Internet Protocol
- PBX: Private Branch Exchange
- SIP: Session Initiation Protocol
- TDM: Time Division Multiplexing
- VoIP: Voice Over IP (Voice Over Internet Protocol)