# Dialogic IP-Media Gateway Installation and Configuration Integration Note

## 1. Scope

This document is intended to detail a typical installation and configuration of a PBX IP-Media Gateway when used to interface between a PBX and a unified messaging type 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

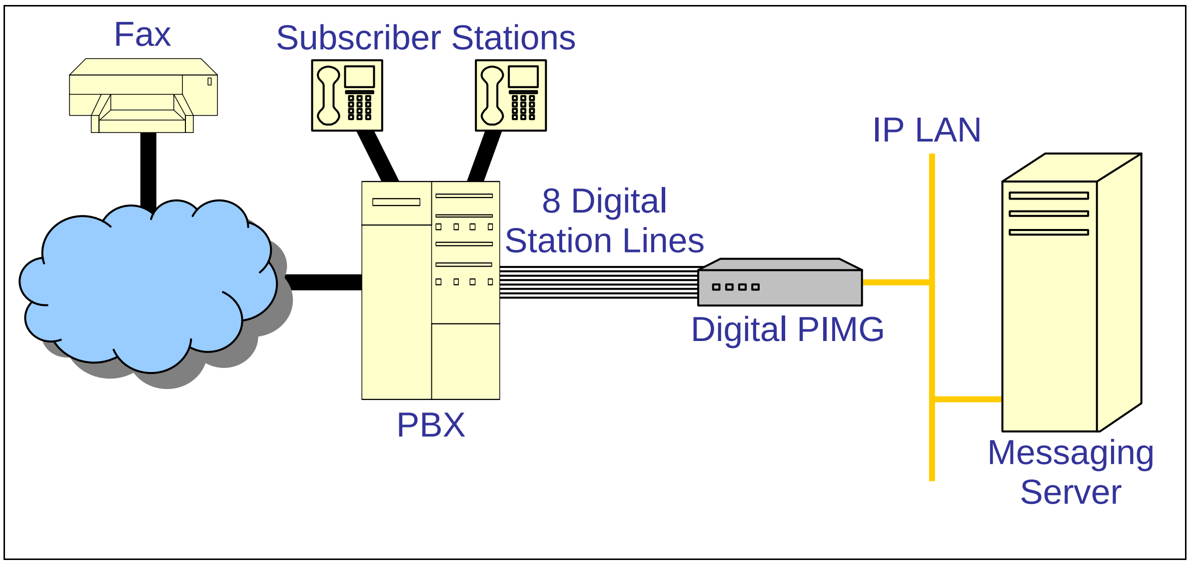
| | |
|---|---|
| PBX Vendor | Nortel |
| Model(s) | Option 11c |
| Software Version(s) | Release 25 |
| Additional Notes | N/A |

### 2.2 Gateway

| | |
|---|---|
| Gateway Model | PIMG80DNI |
| Software Version(s) | 5.0.42 |
| Protocol | Digital set emulation |

### 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 needs to have the following software packages installed

- #19 (DDSP) Digit Display Software
- #46 (MWC) Message waiting software (only required to perform MWIs)

### 3.1.1 PBX Equipment Required

- To support the 2-wire station interface as documented you need the one of the following PBX line card types:
  - NT8D02xx (where xx = revision) – 16 port card
  - NTDK16 - 48 port version built directly into the Option 11c mini

- NOTES
  - Dialogic has been able to certify a wide range of NT8D02 revisions including EA and GA.
  - Option 11c mini PBXs come with a 48 port digital card built in. Additional digital ports can be added via using standard NT8D02 type line cards.
  - Nortel PBXs ranging from the Option 11 up to the Option 81 all use the NT8D02 type line cards.

### 3.1.2 PBX Cabling Requirements

- It is recommended that total loop length (cable distance between PBX connection and the gateway interface) be no longer than 3000 feet (915 meters) and no shorter than 6 feet (2 meters).
- The Nortel digital set interface on the gateway is polarity sensitive.

## 3.2 Gateway Prerequisites

The gateway needs to support m2616 digital station set emulation.

# 4. Summary of Limitations

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

# 5. Gateway Setup Notes

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

- Assign 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.
- Configure the gateway for the M1 integration.

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

- Configure the gateway with at least a single IP endpoint pointing to your messaging server. If multiple IP endpoints are to be sued then ensure they are configured as well.
- Activate fault tolerance and load balancing as required by the application and system requirements.
- Specify the required audio coders and required by the application.
- Set the Hunt group extension number to the extension number you will be using for your primary DN in the PBX programming.

# 6. PBX Setup Notes

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

- Validating the required PBX software packages.
- Setting up each gateway port.
- Setting up the subscribers stations.

The basic commands that you will encounter on the PBX to perform these actions are:

- `LD10; LD11; LD20; LD22`

## 6.1 Validating Required PBX Software Packages

Use the `LD22` command to print out a listing of the all the installed software packages on the PBX. Once the listing scrolls by review it to ensure that all the required packages are included. The example below shows the output of this command (truncated for brevity) and what to look for to locate if your PBX has the proper software loaded.

```
>LD 22
REQ   PRT
TYPE  PKG
OPTF
CDR
CTY
DISA
NCOS
DDSP
MWC
```

Make note of the `DDSP` and `MWC` packages in the list. These are the require packages you must have. If you do not have these packages contact your PBX vendor to have them installed.

## 6.2 Setting Up Each Gateway Station Port

Use the `LD11` command to configure as many station ports as required for your configuration (8 ports per gateway). The example below shows performing this command on one station.

```
>LD 11
REQ   NEW
TYPE  2616
TN    6 0
DES
CUST  0
AOM
FDN
TGAR
NCOS
RNPG
SSU
CLS   ADD CNDA
HUNT  5001
LHK   0
KEY   00 SCR 5000
KEY   13 MIK
KEY   14 MCK
KEY   15 TRN
```

Important notes about the above programming:

1. The REQ field is where you specify the command for this overlay. You can choose to build a new station set using the NEW command or make changes to an existing station using the CHG command.
2. The TYPE field is critical as it specifies the station type configured for this sport. This MUST be set to 2616.
3. The CLS field is required to have both ADD and CNDA configured. ADD forces the display of the station to update as soon as the call rings to it, and CNDA forces the PBX to make sure that reason codes are present in the display when the call arrives.
4. The first key on the station set (KEY 00) must be configured as a SCR (Single Call Ringing) type.
5. The HUNT parameter is used to direct this station to forward to the next station in line under busy conditions. This is the setting that controls the grouping together of station sets in a hunting pattern for call distribution. This example shows station 5000 hunting to station 5001. Station 5001 would be configured to hunt to 5002 and so forth until your hunting pattern is completed at the last port to be built.
6. KEY 13 and KEY 14 must be set to MIK and MCK as shown for MWI functionality to operate properly.
7. KEY 15 must be configured to TRN (transfer) for transfers to operate properly.

## 6.3 Setting Up Subscriber Station Sets

This is an example of how to set up a subscriber that uses a digital station set to forward correctly to the voice messaging server. Use the LD11 command to change the stations parameters as shown below.

```
>LD 11
REQ   CHG
TYPE  2008
TN    0 1 8 3
ECHG
DES
FDN   5000
TGAR
HUNT  5000
NCOS
RNPG
SSU
CLS   HTA FNA MWA CFTA SFA
EFD   5000
EHT   5000
.
.
.
```

Important notes about the above programming:

1. The FDN field is where you specify the destination for this station set to forward to under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
2. The CLS field is required to have HTA, FNA, MWA, CFTA and SFA configured. If these are not configured properly the remainder of the programming is not going to provide you with the proper prompts to continue.
3. The EFD field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
4. The HUNT field is where you specify the destination for internal calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.

5. The EHT field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
6. The MWA setting in the CLS field allows this station to make use of its MWI light. If this is not configured the stations MWI lamp will not work.

This is an example of how to set up a subscriber that uses an analog station set to forward correctly to the voice messaging server. Use the LD10 command to change the stations parameters as shown below.

```
>LD 10
REQ   CHG
TYPE  500
TN    0 0 7 1
CDEN
DES
FDN   5000
CUST
DIG
DN
HUNT  5000
TGAR
NCOS
RNPG
CLS   HTA FNA MWA LPA CFTA SFA
FTR
EFD   5000
EHT   5000
.
.
.
```

Important notes about the above programming:

1. The FDN field is where you specify the destination for this station set to forward to under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
2. The CLS field is required to have HTA, FNA, MWA, LPA, CFTA and SFA configured. If these are not configured properly the remainder of the programming is not going to provide you with the proper prompts to continue.
3. The EFD field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
4. The HUNT field is where you specify the destination for internal calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
5. The EHT field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
6. The MWA setting in the CLS field allows this station to make use of the MWI feature.
7. The LPA setting in the CLS field controls the phones MWI notification method. On analog stations with a neon MWI lamp this setting must be included to use it. Without this setting the analog station will only have stutter dial tone as its notification method.

## 7. 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 call scenarios** | | |
| 1 | Direct call to hunt group. | The calling party number is expected to be contained in the From header of the Invite. |
| 2 | Internal ring-no-answer forward. | The called party will be shown in the Diversion header of the invite. The calling party will be contained in the From header. The reason of the diversion header is shown as no-answer. |
| 3 | External ring-no-answer forward. | The called party will be shown in the Diversion header of the invite. The calling party (if available) will be contained in the From header. The reason of the diversion is shown as no-answer. |
| 4 | Internal busy forward from a subscribers station set. | The called party will be shown in the Diversion header of the invite. The calling party will be contained in the From header. The reason of the diversion header is shown as busy. |
| 5 | External busy forward from a subscribers station set. | The called party will be shown in the Diversion header of the invite. The calling party will be contained in the From header. The reason of the diversion header is shown as busy. |
| 6 | Internal all call forward from a subscribers station set. | The called party will be shown in the Diversion header of the invite. The calling party will be contained in the From header. The reason of the diversion header is shown as fwd-all. |
| 7 | External all call forward from a subscribers station set. | The called party will be shown in the Diversion header of the invite. The calling party will be contained in the From header. The reason of the diversion header is shown as fwd-all. |

| | | |
|---|---|---|
| **Transfer Scenarios** | | |
| 8 | Blind transfer to a station from messaging server where the destination answers the call. | The transfer is completed once the destination is judged as connected. Depending upon the speed that the destination is answered the caller and called parties may be connected together with a slight bit of the called parties voice clipped. |
| 9 | Blind transfer to a station from messaging server where the destination does not answer the call. | If the station is configured to forward back to the gateway then the call will arrive looking as a forwarded call with the called party being the transfer destination but the calling party may be the gateway port performing the transfer, depending on how quickly the transfer to the destination can be completed. |
| 10 | Blind transfer to a subscribers station from messaging server where the destination is busy. | The transfer should fail. |
| 11 | Blind transfer to an invalid number. | The transfer should fail. |
| 12 | Supervised transfer to a subscribers station from messaging server where the user does not answer the call. | The transfer completion speed and timing is up to the application. The application should decide to either complete the transfer and let the stations forwarding carry it back to the gateway or abort it before the forwarding happens. |
| 13 | Supervised transfer to a subscribers station from messaging server where the user answers the call. | The transfer completion speed and timing is up to the application. |
| 13 | Supervised transfer to a subscribers station from messaging server where the destination is busy. | The transfer completion speed and timing is up to the application. The application should decide to either complete the transfer and let the stations forwarding carry it back to the gateway or abort it before the forwarding happens. |
| 14 | Supervised transfer to an Invalid number. | The transfer completion speed and timing is up to the application. |
| **Outbound Call Scenarios** | | |
| 15 | Outbound call to subscriber station that answers. | The call is flagged to the application as completed when the gateway can determine that the call has been connected through. The application should take this into account when making decision when to start the audio stream. |
| 16 | Outbound call to subscriber station that does not answer. | The application needs to take into account if the destination has been set to forward back |

| | | |
|---|---|---|
| | | to the gateway for a ring no answer condition and judge accordingly when to either stop waiting for an answer and cancel the call or know that it will end up arriving back to the gateway as a forwarded call. |
| 17 | Outbound call to subscriber station that is busy. | The application needs to take into account if the destination has been set to forward back to the gateway for a ring no answer condition and judge accordingly when to either cancel the call or know that it will end up arriving back to the gateway as a forwarded call. |
| 18 | Outbound call to an external number. | Depending on the state of the destination the call will either be judged as connected or fail do to busy or error tone conditions. |
| **MWI Scenarios** | | |
| 19 | Turn a subscribers light on that is currently off. | This should return success. |
| 20 | Turn a subscribers light on that is currently on. | This will return a failure as the PBX rejects the request but the gateway is unable to determine if the rejection is the result of an invalid extension or because the far end light is already on. The application needs to make judgment based upon known remote MWI state. |
| 21 | Turn a subscribers light off that is currently on. | This should return success. |
| 22 | Turn a subscribers light off that is currently off. | This should return success. |

# 8. Troubleshooting

## 8.1 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.

## 8.2 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…)

These keys are helpful during specific issues and can be enabled for targeted troubleshooting of very specific cases. Activation of these keys may generate large amounts of data on busy systems and increase the size of the collected log files, but will not harm system performance.

- `dspcpi` (all keys) – This allows the collection of tone related data. This data is very helpful in cases where you think you have problems detection specific tones that should be, should not be, or are expected to be present at specific times during the call. If you do not suspect a tone related issues this key may be left disabled. This data is very important in the following scenarios:
    - Failing transfers
    - Failing outbound calls (play to phone)
    - Dropped calls (callers cut off while leaving messages, etc…)
- `adept` (all keys) – This allows the collection of rule parsing data. This data is required in the troubleshooting of all integration related issues.

*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.*

# 9. Appendix

## 9.1 Abbreviations

| | |
|---|---|
| LBRC | Low Bit Rate Coder |
| MWI | Message Waiting Indication |
| PBX | Private Branch Exchange |

**For more details, go to www.dialogic.com.**

Dialogic Corporation
9800 Cavendish Blvd., 5th floor
Montreal, Quebec
CANADA H4M 2V9

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