Dialogic® Media Gateway Installation and Configuration Integration Note

1. Scope

This document is intended to detail a typical installation and configuration of a Dialogic® Media Gateway when used to interface between a PBX and a unified messaging application.

2. Configuration Details

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

2.1 PBX

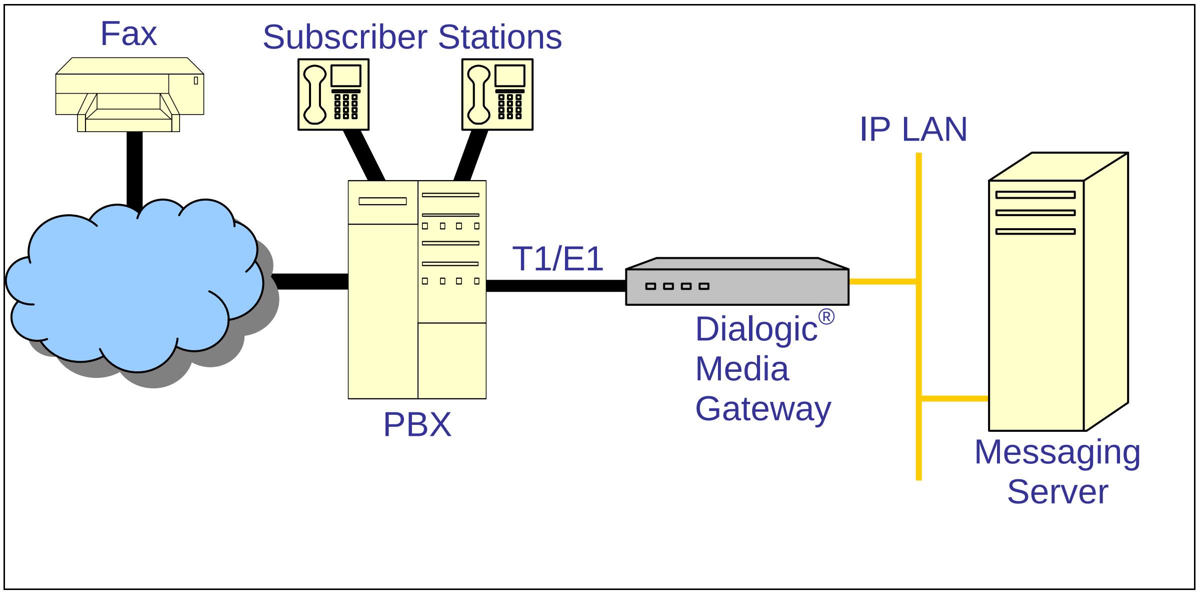
PBX Vendor	Aastra (former Ericsson)
Model(s)	MD110
Software Version(s)	MX1 TSW R2A (BC13)
Additional Notes	See PBX Prerequisites (Section 3.1)

2.2 Dialogic® Gateway

Gateway Model	Dialogic® DMG2xxxDTI (former TIMGxxxDTI)
Software Version(s)	5.1.108
Protocol	E1 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 PBX Equipment and Cabling installed for the QSIG protocol to operate properly and provide all advanced supplemental services.

3.1.1 PBX Equipment Required

To connect to the PBX using E1 QSIG, you need the TLU 76 (ISDN DTI/PRI 2.0) line card.

3.1.2 PBX Cabling Requirements

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

3.2 Gateway Prerequisites

The gateway needs to support E1 QSIG interface.

4. Summary of Limitations

MWI support is currently not implemented from the gateway over the E1 QSIG Trunk. Call Diversion from the gateway back to the PBX is partially supported.

5. Gateway Setup Notes

During the initial setup of the 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.
- Set the Line Mode to E1.
- Set the Protocol to ISDN - QSIG.

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

- Configure the gateway with at least a single IP endpoint pointing to your voice server.
- Set the Voice coder to be either G.711 (default) or G.273 if required.
- Set the Line Encoding and Line Framing as required by your E1 Interface. Typical settings are Encoding = HDB3 and Framing = CRC_MF.

6. PBX Setup Notes

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

- Initiating route category.
- Initiating route data.
- Initiating route equipment.
- Initiating external destination route data.
- Initiating number analysis.
- Configuring application system parameters.
- Configuring call diversion for subscriber stations.

All PBX programming is done via a serial terminal connected to the PBX administration port.

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

Initiating Route Category.	ROCAI
Initiating Route Data.	RODAI
Initiating Route Equipment.	ROEQI
Initiating External Destination Route Data.	RODDI
Initiating Number Analysis.	NANSI
Configuring Application System Parameters.	ASPAC
Configuring Call Diversion for Subscribers.	CDINI

6.1 Initiating Route Category

Initiate the E1 route category using the command ROCAI. Several of the fields that require site-specific entries are:

- `ROU` requires an open route number for the E1 board to use. The command ROCAP:ROU=ALL; will print all used ROU numbers; select any available number from 1-250. For this example, 8 was selected.

The fields of this command which must be modified in this step are:

`ROU, SEL, SERV, TRAF, SIG, BCAP.`

The programming example below shows how to initiate the E1 route category using `ROCAI`. To print the results, use the command **ROCAP**.

```
<ROCAI:ROU=8, SEL=7130000000000010, SERV=2110000001, TRAF=03151515, SIG=511100000031,
BCAP=111111;
<ROCAP:ROU=8;
ROUTE CATEGORY DATA
ROU SEL              TRM SERV       NODG DIST DISL TRAF     SIG          BCAP
8   7130000000000010 5   2110000001  0    5    128  03151515 511100000031 111111

END
```

- At the prompt < enter `ROCAI:ROU=X, SEL=7130000000000010, SERV=2110000001, TRAF=03151515, SIG=511100000031, BCAP=111111;` press `RETURN`.
    - where `X` is the open ROU number to use for the E1 route.

6.2 Initiating Route Data

Initiate the E1 route data using the command `RODAI`. Several of the fields that require site-specific entries are:

- `ROU` requires the ROU number for the E1 board selected previously.

The fields of this command that must be modified in this step are:

`ROU, TYPE, VARC, VARI, VARO.`

The programming example below shows how to initiate route data for the E1 trunk using RODAI. To print the results, use the command RODAP.

```
<RODAI:ROU=8, TYPE=SL60, VARC=00200070, VARI=75540000, VARO=06300000;
<RODAP:ROU=8;
ROUTE DATA
ROU   TYPE  VARC        VARI        VARO        FILTER
8     SL60  H'00200070  H'75540000  H'06300000  NO

END
```

- At the prompt < enter RODAI:ROU=X, TYPE=SL60, VARC=00200070, VARI=75540000, VARO=06300000; press RETURN.
    - where X is the ROU number for the E1 board selected previously.

6.3 Initiating Route Equipment

Initiate the route equipment of the E1 board using command ROEQI. Several of the fields that require site-specific entries are:

- ROU requires the ROU number for the E1 board selected previously.
- TRU requires trunk number, where the first 3 digits are the LIM number, and the last 2 are the channel number.
- EQU requires the equipment position number for the E1 board.

The fields of this command that must be modified in this step are:

ROU, TRU, EQU.

The programming example below shows how to initiate the route equipment for the E1 trunk using ROEQI. To print the results, use the command **ROEDP**.

```
<ROEQI:ROU=8, TRU=001-1&&001-15, EQU=001-0-40-1&&001-0-40-15;
<ROEQI:ROU=8, TRU=001-17&&001-31, EQU=001-0-40-17&&001-0-40-31;
<ROEDP:ROU=8,TRU=ALL;
ROUTE EQUIPMENT DATA
ROU   TRU      EQU           IP ADDRESS          SQU       INDDAT           CNTRL
8     001-1    001-0-40-01                                 H'000000000000
8     001-2    001-0-40-02                                 H'000000000000
8     001-3    001-0-40-03                                 H'000000000000
8     001-4    001-0-40-04                                 H'000000000000
8     001-5    001-0-40-05                                 H'000000000000
8     001-6    001-0-40-06                                 H'000000000000
8     001-7    001-0-40-07                                 H'000000000000
8     001-8    001-0-40-08                                 H'000000000000
8     001-9    001-0-40-09                                 H'000000000000
8     001-10   001-0-40-10                                 H'000000000000
8     001-11   001-0-40-11                                 H'000000000000
8     001-12   001-0-40-12                                 H'000000000000
8     001-13   001-0-40-13                                 H'000000000000
8     001-14   001-0-40-14                                 H'000000000000
8     001-15   001-0-40-15                                 H'000000000000
8     001-17   001-0-40-17                                 H'000000000000
8     001-18   001-0-40-18                                 H'000000000000
8     001-19   001-0-40-19                                 H'000000000000
8     001-20   001-0-40-20                                 H'000000000000
8     001-21   001-0-40-21                                 H'000000000000
8     001-22   001-0-40-22                                 H'000000000000
8     001-23   001-0-40-23                                 H'000000000000
8     001-24   001-0-40-24                                 H'000000000000
8     001-25   001-0-40-25                                 H'000000000000
8     001-26   001-0-40-26                                 H'000000000000
8     001-27   001-0-40-27                                 H'000000000000
8     001-28   001-0-40-28                                 H'000000000000
8     001-29   001-0-40-29                                 H'000000000000
8     001-30   001-0-40-30                                 H'000000000000
8     001-31   001-0-40-31                                 H'000000000000

END
```

- At the prompt > enter ROEQI:ROU=X, TRU=YYY-1&&YYY-15, EQU=YYY-M-ZZ-1&&YYY-M-ZZ-15; press RETURN.
  - where X is the ROU number for the E1 board selected previously.
  - where YYY is the LIM Number for the E1 board.
  - where M is the MAG/DSU number for the E1 board.
  - where ZZ is the Board Position number for the E1 board.
- At the prompt > enter ROEQI:ROU=X, TRU=YYY-17&&YYY-31, EQU=YYY-M-ZZ-17&&YYY-M-ZZ-31; press RETURN.
  - where X is the ROU number for the E1 board selected previously.
  - where YYY is the LIM Number for the E1 board.
  - where M is the MAG/DSU number for the E1 board.
  - where ZZ is the Board Position number for the E1 board.

6.4 Initiating External Destination Route Data

Initiate the External Destination Route Data using command RODDI. Several of the fields that require site-specific entries are:

- DEST requires an unused number in the dial plan.
- ROU requires the ROU number for the E1 board selected previously.

The fields of this command that must be modified in this step are:

DEST, ROU, ADC.

The programming example below shows how to initiate the external destination route data using RODDI. To print the results, use the command **RODDP**.

```
<RODDI:DEST=81, ROU=8, ADC=06062000000002501060000001;
<RODDP:DEST=81;
EXTERNAL DESTINATION ROUTE DATA
DEST   DRN ROU CHO CUST ADC                        TRC SRT NUMACK PRE
81         8            06062000000002501060000001 0   1   0

END
```

- At the prompt < enter RODDI:DEST=YY, ROU=X, ADC=06062000000002501060000001; press RETURN.
  - where X is the ROU number for the E1 board selected previously.
  - where YY is the DEST number chosen to route calls to the E1 Trunk.

6.5 Initiating Number Analysis

6.5.1 E1 Trunk Destination Number

Now that the destination number is assigned, it must be added to the PBX Number analysis using the command **NANSI**. Several of the fields that require site-specific entries are:

- NUMSE requires the DEST number assigned to the E1 trunk previously.

The fields of this command that must be modified in this step are:

NUMTYP, NUMSE.

The programming example below shows how to add the trunk number series to the PBX Number Analysis using NANSI. To print the results, use the command **NADAP**.

```
<NANSI:NUMTYP=ED, NUMSE=81;
<NADAP:NUMTYP=ED;
NUMBER ANALYSIS DATA
TYPE OF SERIES                  NUMBER SERIES
EXTERNAL DESTINATION CODE          81
```

- At the prompt < enter NANSI:NUMTYP=ED, NUMSE=YY; press RETURN.
  - where YY is the DEST number to route calls to the E1 Trunk selected previously.

6.5.2 PBX Own Exchange Destination Number

In order to handle Path Replacement on Join Transfer and Call Redirection, the PBX must have an Own Exchange Number Series assigned to route calls back to itself. The Own Exchange Number must be added to the PBX Number Analysis using the command **NANSI**. Several of the fields that require site-specific entries are:

- NUMSE requires an unused number in the dial plan for the Own Exchange Number.

The fields of this command that must be modified in this step are:

`NUMTYP, NUMSE.`

The programming example below shows how to add the own exchange number series to the PBX Number Analysis using `NANSI`. To print the results, use the command `NADAP`.

```
<NANSI:NUMTYP=EN, NUMSE=80;
<NADAP:NUMTYP=EN;
NUMBER ANALYSIS DATA
TYPE OF SERIES                    NUMBER SERIES
OWN EXCHANGE NUMBER SERIES           80
```

- At the prompt `<` enter `NANSI:NUMTYP=EN, NUMSE=XX;` press `RETURN.`
    - where `XX` is the unused Number Series number to identify the PBX in the private network.

6.6 Configuring Application System Parameters

Configure the Application System Parameters using the command `ASPAC.`

The fields of this command that must be modified in this step are:

`PARNUM, PARVAL.`

The Values of the Application System Parameters that must be modified are:

- `PARNUM=44` Rerouting on no reply on a call to private external line
- `PARNUM=66` Route optimization availability
- `PARNUM=70` Time before route optimization starts on alternative routing
- `PARNUM=71` Time before route optimization starts on transfer
- `PARNUM=72` Time before restart of route optimization when request denied
- `PARNUM=73` Attempts on route optimization when the request denied
- `PARNUM=77` Traffic category check at diversion
- `PARNUM=78` Traffic category check at diversion on no answer
- `PARNUM=79` Extension's permission to dial message diversion service codes
- `PARNUM=85` Rerouting incoming call before complete internal number received
- `PARNUM=93` ISDN call diversion mode
- `PARNUM=98` Automatic activation of diversion on busy
- `PARNUM=105` Automatic activation of diversion on no answer
- `PARNUM=156` Call discrimination check for Deflect/SST case
- `PARNUM=223` Type of network services

The programming example below shows how to configure the application system parameters using ASPAC. To print the results, use the command ASPAP.

```
<ASPAC:PARNUM=223, PARVAL=7;
<ASPAP:PARNUM=223;
APPLICATION SYSTEM PARAMETERS
PARNUM      PARVAL
   223           7
END
```

- At the prompt < enter ASPAC:PARNUM=XX,PARVAL=YY; press RETURN.
    - where XX is the Parameter Number.
    - where YY is the Value to be assigned to the Parameter, as defined below.
- Repeat for each PARNUM and PARVAL combination below:
    - PARNUM=44 PARVAL=3
    - PARNUM=66 PARVAL=1
    - PARNUM=70 PARVAL=1
    - PARNUM=71 PARVAL=1
    - PARNUM=72 PARVAL=1
    - PARNUM=73 PARVAL=3
    - PARNUM=77 PARVAL=0
    - PARNUM=78 PARVAL=0
    - PARNUM=79 PARVAL=1
    - PARNUM=85 PARVAL=1
    - PARNUM=93 PARVAL=0
    - PARNUM=98 PARVAL=1
    - PARNUM=105 PARVAL=1
    - PARNUM=156 PARVAL=0
    - PARNUM=223 PARVAL=7

6.7 Configuring Call Diversion for Subscriber Stations

Configure call forwarding for individual subscriber to the E1 Trunk using the command CDINI. Several of the fields that require site-specific entries are:

- DIR requires the directory number in the dial plan for the Subscriber Station.
- DIV requires the DEST number of the E1 trunk assigned previously.

The fields of this command that must be modified in this step are:

DIR, DIV.

The programming example below shows how to configure a diversion destination for a subscriber using CDINI. To print the results, use the command CDIDP.

```
<CDINI:DIR=1017,DIV=81;
<CDIDP:DIR=1017;
CALL DIVERSION INDIVIDUAL DATA
DIR          DIV
1017         81

END
```

- At the prompt < enter CDINI:DIR=XXXX,DIV=YY; press RETURN.
    - where XXXX is the directory number defined for the subscriber station.
    - where YY is the DEST number of the E1 Trunk previously assigned.

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 subscriber's 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 subscriber's 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 subscriber's 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 subscriber's 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. The calling party is not available. The transfer was by join, not a path replacement.
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. The calling party is not available. The transfer was by join, not a path replacement.
10	Blind transfer to a subscriber's 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 subscriber's 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. The calling party is not available. The transfer was by join, not a path replacement.
13	Supervised transfer to a subscriber's station from messaging server where the user answers the call.	The transfer completion speed and timing is up to the application. The calling party is not available. The transfer was by join, not a path replacement.
13	Supervised transfer to a subscriber's 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. The calling party is not available. The transfer was by join, not a path replacement.

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 subscriber's light on that is currently off.	This should return success.
20	Turn a subscriber's light on that is currently on.	This should return success.
21	Turn a subscriber's light off that is currently on.	This should return success.
22	Turn a subscriber's 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 troubleshooting scenarios and should be considered keys to activate by default for 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 circuit-side activity of the emulated station set, such as display updates, key presses, light transitions and hook state changes. This data is 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 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 should not harm system performance.

- `dspif` (all keys) – This allows the collection of tone-related data. This data is 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 issue, this key may be left disabled.

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

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