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

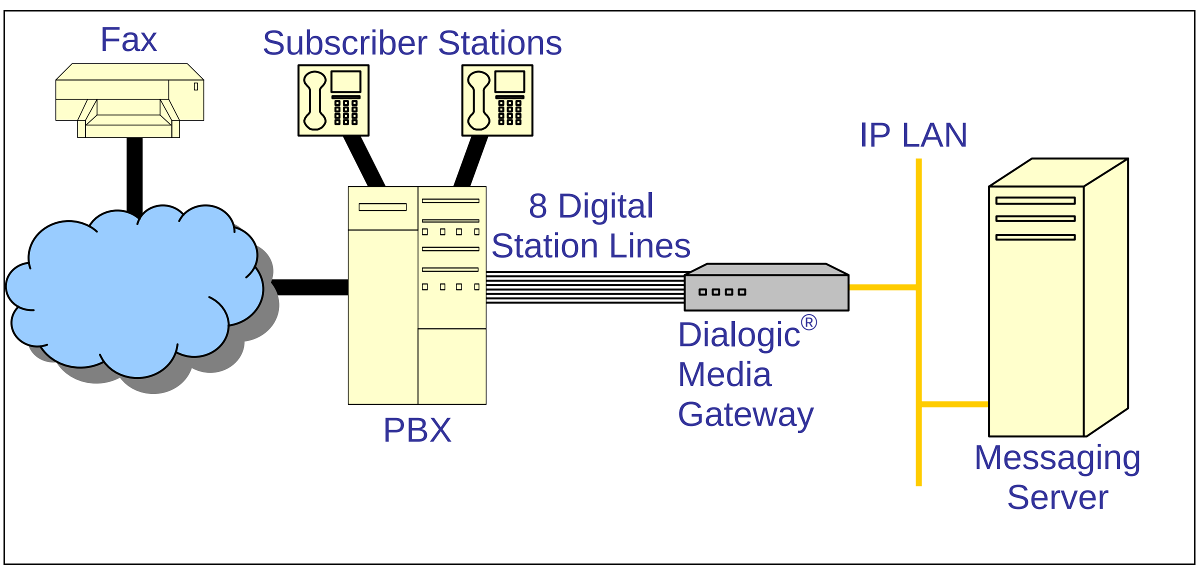
PBX Vendor	Aastra (former Ericsson)
Model(s)	MD110
Software Version(s)	MX1 TSW R2A (aka BC13)
Additional Notes	N/A

2.2 Dialogic® Gateway

Gateway Model	Dialogic® DMG1008LS (former PIMG80LS)
Software Version(s)	5.1.10
Protocol	Serial MD110

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

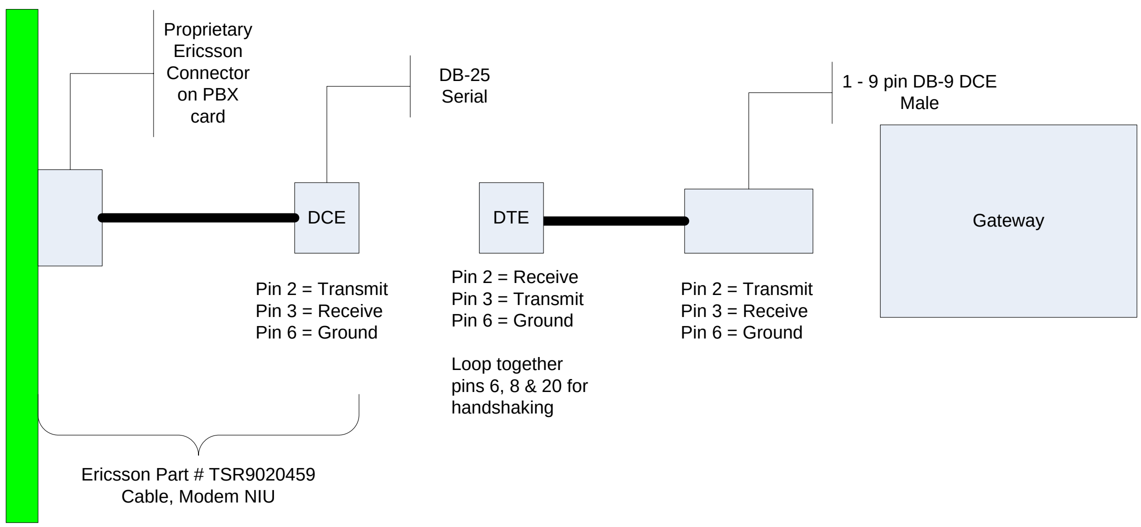
The MD110 needs to be enabled / licensed for serial SMDI.

3.1.1 PBX Equipment Required

- To support the 2-wire analog station interface, you need the ELU-29 analog PBX line card.
- To support the MD110 serial interface on the PBX, you will need a TSR9020459 serial cable to interface to the PBX via serial MD110 integration.

3.1.2 PBX Cabling Requirements

Because the integration data is transmitted along a secondary serial data path between the PBX and the gateway, it will be required to either purchase or construct a cable that connects the serial port of the Ericsson PBX to the serial port of the gateway. The diagram below provides details on how this cable was constructed for the creation of this document.



The industry rule of thumb for RS232 serial communications is to keep the length of the cable no longer than 50 feet. Lengths longer than the recommended standard may need additional equipment to ensure correct transmission levels are maintained.

3.2 Gateway Prerequisites

N/A

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 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.

NOTE: The analog gateway is an interface specific SKU (only analog FXO) and therefore does not require being told what specific PBX it is being configured for.

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

- Configure the gateway with at least a single IP endpoint.
- Set the Voice coder to be either G.711 (default) or G.273 if required.
- Set the Serial Mode to Master for the single master gateway and to Slave for any of the additional slave gateways (if you are using any slaves).
- Ensure that the serial port settings (baud rate, parity and stop bits) match the port settings on the PBX.
- Set the Serial Interface Protocol to MD110.
- Set the system number to the value provided to you by your PBX administrator.
- Set the Voice mail port length to the length of your LTN numbers provided by the PBX. The typical value for this is 3 since the LTN numbers typically appear formatted as 3-digit, zero padded values like '001'.
- Set each of the Logical Extension Numbers to the terminal numbers provided by the PBX. These values typically start with 1 and advance upwards.

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:

- Setting up serial integration interface.
- Enabling MWI on system.
- Setting up each gateway station port.
- Defining hunt group to act as a central point for incoming calls to the gateway.
- Setting up subscriber station sets.

All PBX programming is done via a web-based interface accessible via a web browser.

6.1 Setting Up Serial Integration Interface

Use the `IO Port Initiate` (`IOEQI`) command to prepare the system to use a specific IO port for the serial integration traffic. An example of this is shown below:

```
IOEQI:IODEV=VMAL,EQU=1-0-60-02,USAGE=OUT;
```

Important notes about the above programming:

1. The `IODEV` setting should always be set to `VMAL`.
2. The `EQU` setting will point to the position of the port on the PBX.
3. The `USAGE` setting should be set to `OUT` or the port will not function as an integration port.

Once the port has been initiated you need to configure it with the proper serial settings to communicate to the gateway.

Use the `Information Computer Function Initiate` (`ICFUI`) command to configure the ports parameters. An example of this is shown below:

```
ICFUI:IFCIND=0,EQU=1-0-60-
2,RATE=2400,DFMT=4,UPDFCN=YES,PARTY=EVEN,CCHECK=YES,TXC=N0,FILLER=62;
```

Important notes about the above programming:

1. The `EQU` setting will point to the position of the port in the PBX where your integration port is located. This is the same number as pointed to by the `IOEQI` setting above.
2. The `RATE` setting should be set to a value that matches what you plan to use on the gateway as a baud rate. A value of `2400` is typical here.
3. The `DFMNT` setting controls the length of your directory numbers.
4. The `PARITY` setting needs to match the parity setting you select on the gateway.
5. The `UPDFCN` setting should be set to `YES`.

6.2 Enabling MWI on System

The next step is to enable message waiting lights for the system. To do this use the `Information Computer Function Change` (`ICFUC`) command. An example of this is shown below:

```
ICFUC:MWF=ALL;
```

6.3 Setting Up Each Gateway Station Port

Use `the Extension Initiate` (`EXTEI`) command to build one analog extension for each of your gateway ports (8 per PIMG). An example of this is shown below:

```
EXTEI :DIR=2011,TYPE=EL6,EQU=1-0-41-2,TRAF=00151501,SERV=000212070,
CDIV=000071121,ROC=020001,ICAT=0004;
```

Important notes about the above programming:

1. The `DIR` setting is the extension number of the port you are configuring.
2. The `TYPE` setting is an equipment type setting. This should be set to `EL6`
3. The `EQU` setting will point to the position of the port on the PBX.
4. The `TRAF`, `SERV`, `CDIV` and `ROC` settings can be site specific as they deal with various service and routing categories. Consult he local PBX admin about these values.
5. The `ICAT` setting should be set to `0004`. This setting indicates the following parameters for the `EL6` type.
    a. Speech services
    b. No polarity reversal
    c. Normal tones
    d. Loop current disconnection (remote disconnect notification)

Build one port for each of the gateway ports you will be using.

6.4 Defining Hunt Group

Use the `Voice Mail Port Initiate` (`VMPOI`) command to build a hunt group and add all the gateway ports. An example of this is shown below:

```
VMPOI:IFCIND=0,PORT=2011&2012&2012&2013&2014&2015&2016&2017,GRP=1050
```

Important notes about the above programming:

1. The `PORT` setting is a listing of the extensions to include in the group. You can use `&` to build a list if your ports are not contiguous or you can use `&&` to provide a starting point and an ending point if all your gateway ports are contiguous.
2. The `GRP` setting gives the group an extension number to use as a forwarding point for subscriber stations and as an access number to call directly into the voice processing system.

Once the hunt group is built use the `Voice Mail Function Initiate` (`VMFUI`) command to turn on the voice mail integration functionality. An example of this is shown below:

```
VMFUI:IFCIND=0,VMF=EXTN2,POFMT=3;
```

Important notes about the above programming:

1. The `VMF` setting controls the registered functionality for the integration port and should be set to `EXTN2`.
2. The `POFMT` setting denotes how many digits the LTN numbers for each voice mail port are going to be. The example above is setting them to 3 digits so when a call comes to the first port the serial data packet is going to list the LTN as '001'. This has an impact on the voice mail port length settings in the gateway as they need to match.

6.5 Setting Up Subscriber Station Sets

All the forwarding of the subscriber station sets can be defined directly on subscriber station set using feature access codes. The subscriber should be directed to set their internal and external ring no answer and busy forwarding conditions to the extension number assigned to the hunt group configuration.

6.6 Additional Comments

N/A

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.

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 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 call that gets forwarded back to the messaging server will contain the calling party listed as the gateway port performing the transfer, not the party that requested the transfer or the original calling party.
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.
13	Supervised transfer to a subscriber's station from messaging server where the destination is busy.	The call that gets forwarded back to the messaging server will contain the calling party listed as the gateway port performing the transfer, not the party that requested the transfer or the original calling party.
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.
- HyperTerminal – Used to test the output of the PBXs serial port to determine if data is being sent by the PBX. This helps in the validation of the cabling and all connects on the serial interface between the PBX and the gateway.
- RS232 breakout box – This tool can be valuable in doing serial cabling work. It helps you determine what signals are available and if data is being either transmitted or received.

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…)

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.

- `dspcpi` (all keys) – This allows the collection of tone-related data. This data is helpful in cases where you think you have problems detecting 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. This data is important in the following scenarios:
  - Failing transfers
  - Failing outbound calls (play to phone)
  - Dropped calls (callers cut off while leaving messages, etc…)
- `si` – This allows the collection of all inbound and outbound serial data on the serial master gateway. This data is required in the troubleshooting of all integration-related issues seen on the gateway designated as the serial master.
- `siip` – This allows the collection of all inbound and outbound serial data on any of the serial slave gateways. This data is required in the troubleshooting of all integration-related issues seen on the gateway designated as a serial slave.
- `simwi` – This allows the collection of all activity reacted to the processing of MWIs using the serial port. This data is required for any MWI-related issues while using the serial interface.

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
FXO	Foreign Exchange Office
CPID	Call Party Identification

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