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.
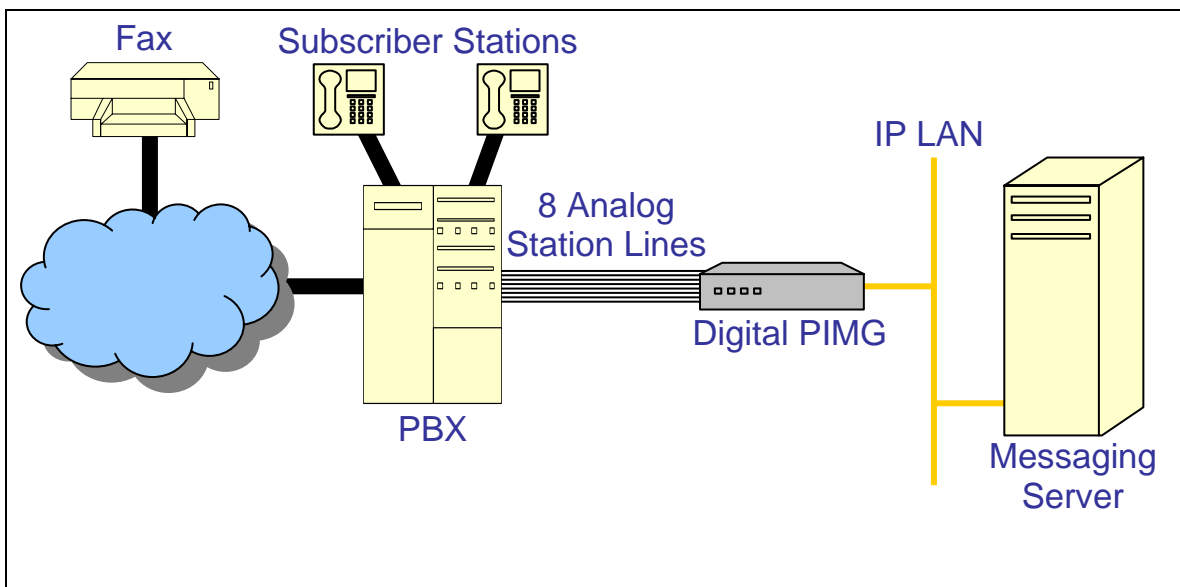
2.1 PBX

PBX Vendor	Toshiba
Model(s)	CTX
Software Version(s)	AR1ME021.00
Additional Notes	N/A

2.2 Gateway

Gateway Model	PIMG80LS
Software Version(s)	5.1.70
Protocol	Analog In band DTMF tones

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

N/A

3.1.1 PBX Equipment Required

- To support the 2-wire analog station interface you need the RSTU3 analog PBX line card.

3.1.2 PBX Cabling Requirements

N/A

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 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.

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 Dialogic gateway using the web interface you must:

- Configure the gateway with at least a single IP endpoint.
- Set the Voice coder to be either G7.11 (default) or G.273 if required.

5.1 Parsing rules

In order to correctly judge remote disconnection of TDM side devices the gateway CPID Parser needs to be configured to recognize the digit sent to it from the PBX to signal disconnection. The setup details below show the PBX is configured to send a DTNF 'D' tone to the gateway. For the gateway to correctly judge this tone as a disconnection tone we need to set up a parsing rule to handle it.

```
# D Disconnect string
rule D
reason disconnect

#direct call to hunt group
rule 94\d*
src_number 1
reason direct
origin internal

#forward-no-answer call
rule 93\d*
dst_number 1
reason no-answer
origin internal

#busy-forwarded call
rule 92\d*
dst_number 1
reason busy

#fwd-all call
rule 91\d*
dst_number 1
reason fwd-all
```

These rules can be edited directly into the Analog web page.

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 the analog Inband integration.
- Setting up each gateway port.
- Defining a hunt group to act as a central point for incoming calls to the gateway.
- Setting up the subscribers stations.

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

6.1   Setting Up Inband integration

Use the `System Voice Mail Data` menu selection to configure the system data settings required for the system to provide the Inband DTMF integration data. An example of this is shown below:
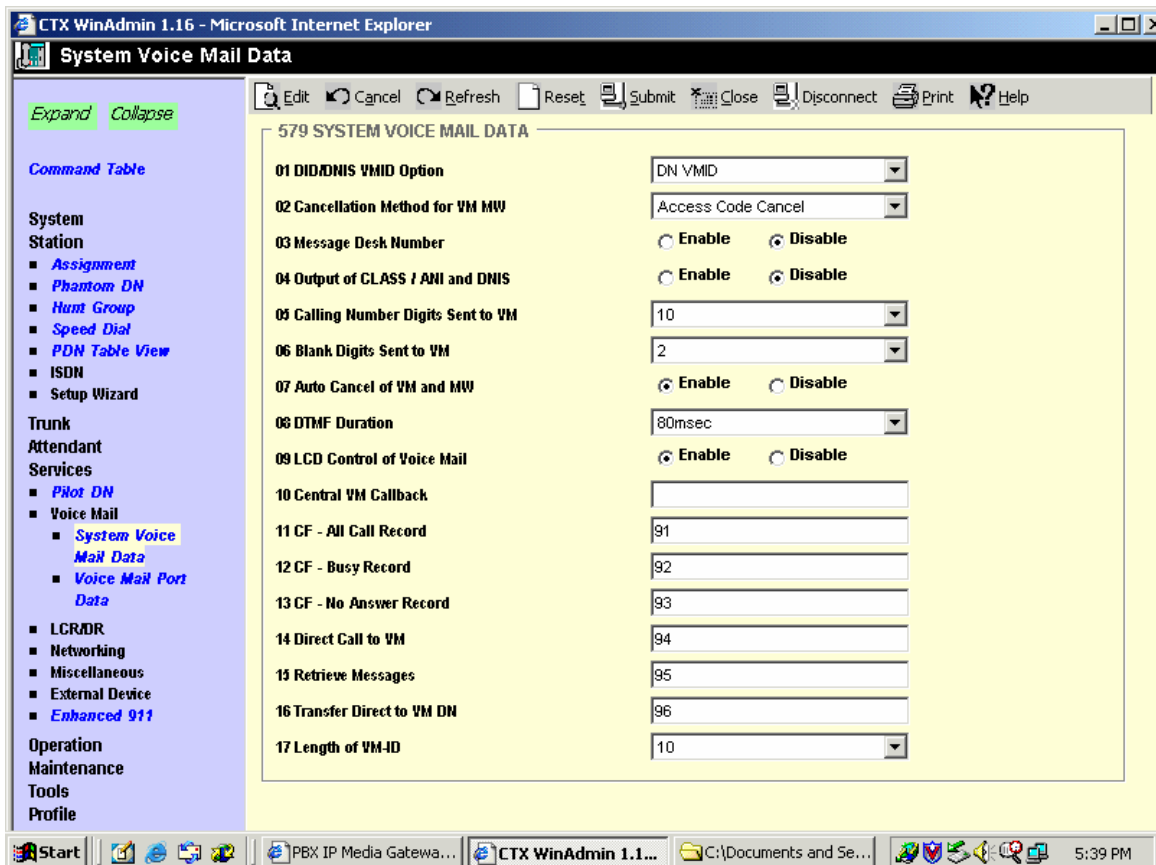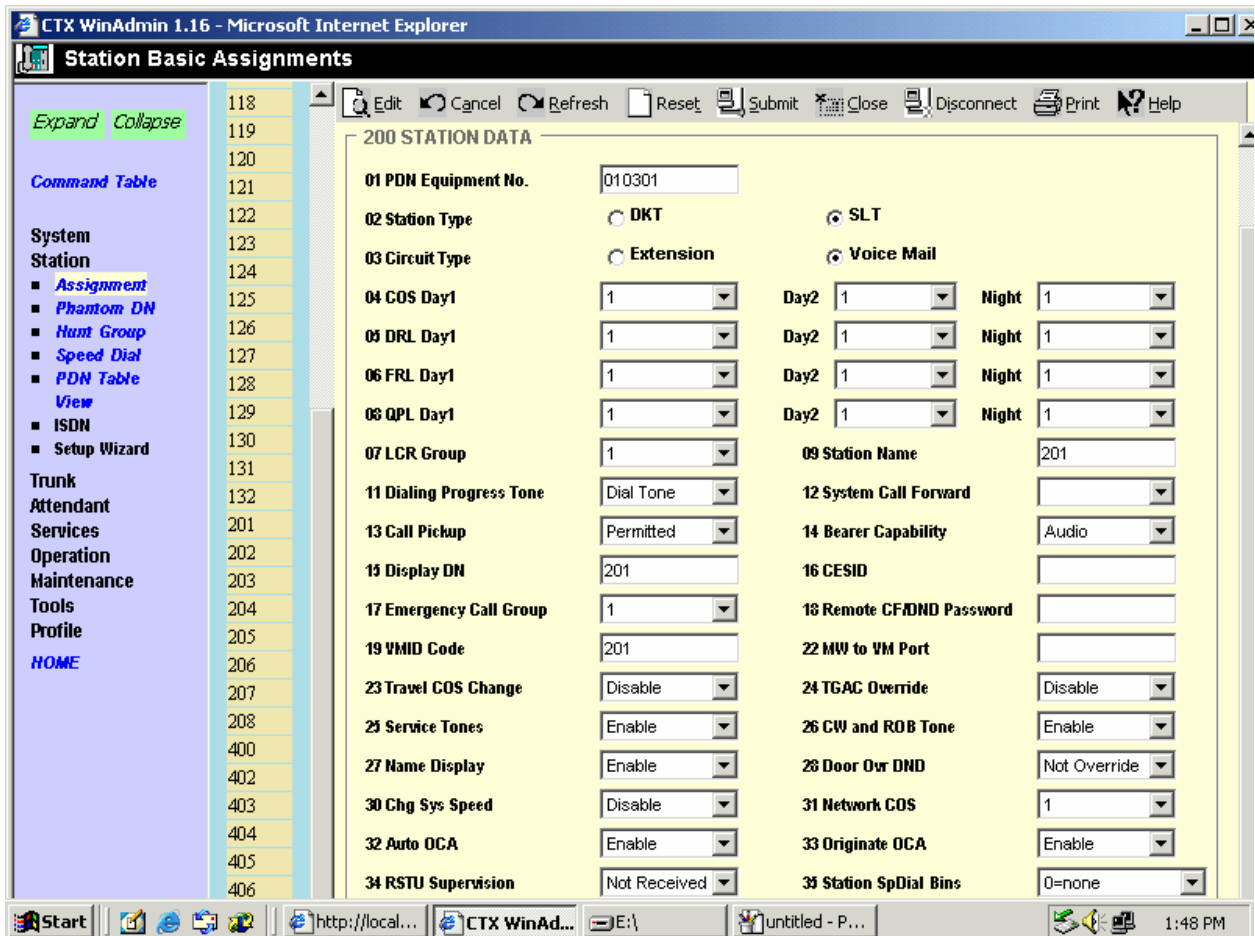

Important notes about the above programming:

1. The entries for settings 11 – 14 need to be configured for unique values as this data will be sent to the gateway as part of the Inband DTMF digit stream. These entries are used to help determine the different call types (busy, ring no answer, all call forwarded, etc…) and need to be unique to allow the parsing rules operate properly.

6.2   Setting Up Each Gateway Station Port

Use the `System Class of Service` menu selection to set up a class of service temple that will be shared across all the gateway station ports. The example below shows performing the command.
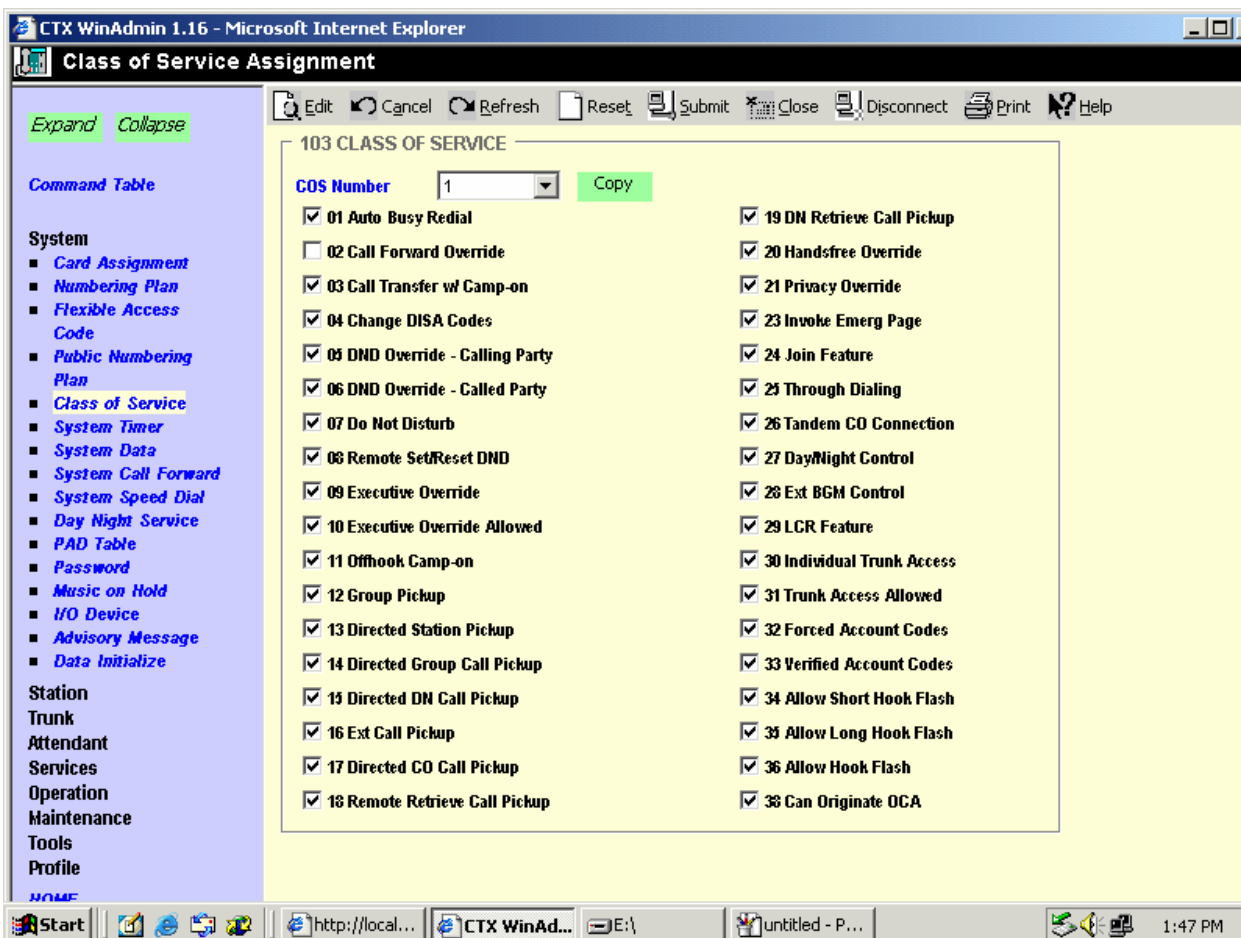

Use the `Station Assignment` menu selection to build as many analog station ports as required for your configuration (8 ports per gateway). This command is what builds the physical interface on the PBX. The example below shows performing this command on one station.

Important notes about the above programming:

1. The `PDN Equipment No.` field is the physical wiring address of the port on the PBX.
2. The `Station Type` field should be set to `SLT`.
3. The `Circuit Type` field should be set to `Voice Mail`.
4. All the class of service (COS) fields should be set to use the class of service defined in the previous step.
5. The `Dialing Progress Tone` field should be set to `Dial Tone`.
6. The `Station Name` field, `VMID Code` field, and `Display DN` field should be configured for the same number are is where you specify the extension number for each analog station port.
7. The `Service Tones` field should be set to Enabled.

In addition to the physical definition you need to tell the PBX that each port is part of a voice mail type and how to present the integration data to the automated system. This is done by using the `Voice Mail Port Data` menu selection. This should be done for each of the analog station ports you defined to use with the gateway. The example below shows performing this command on one station.
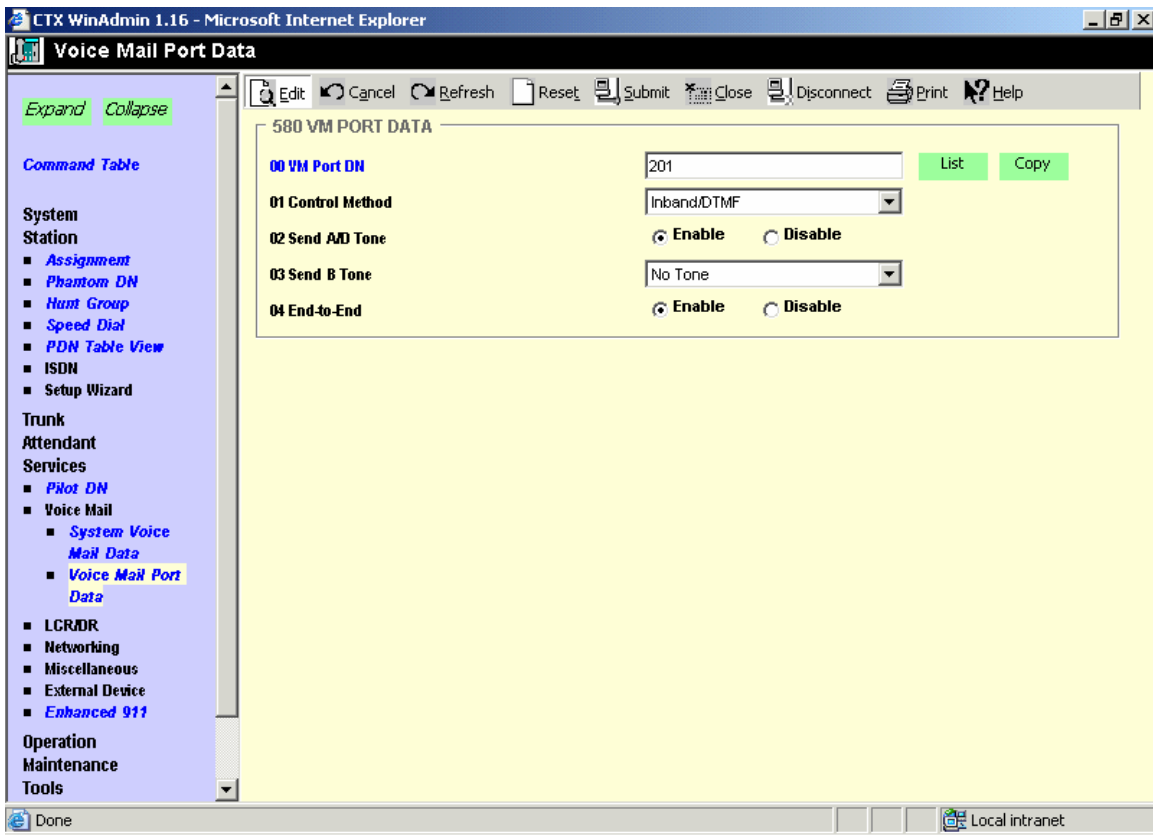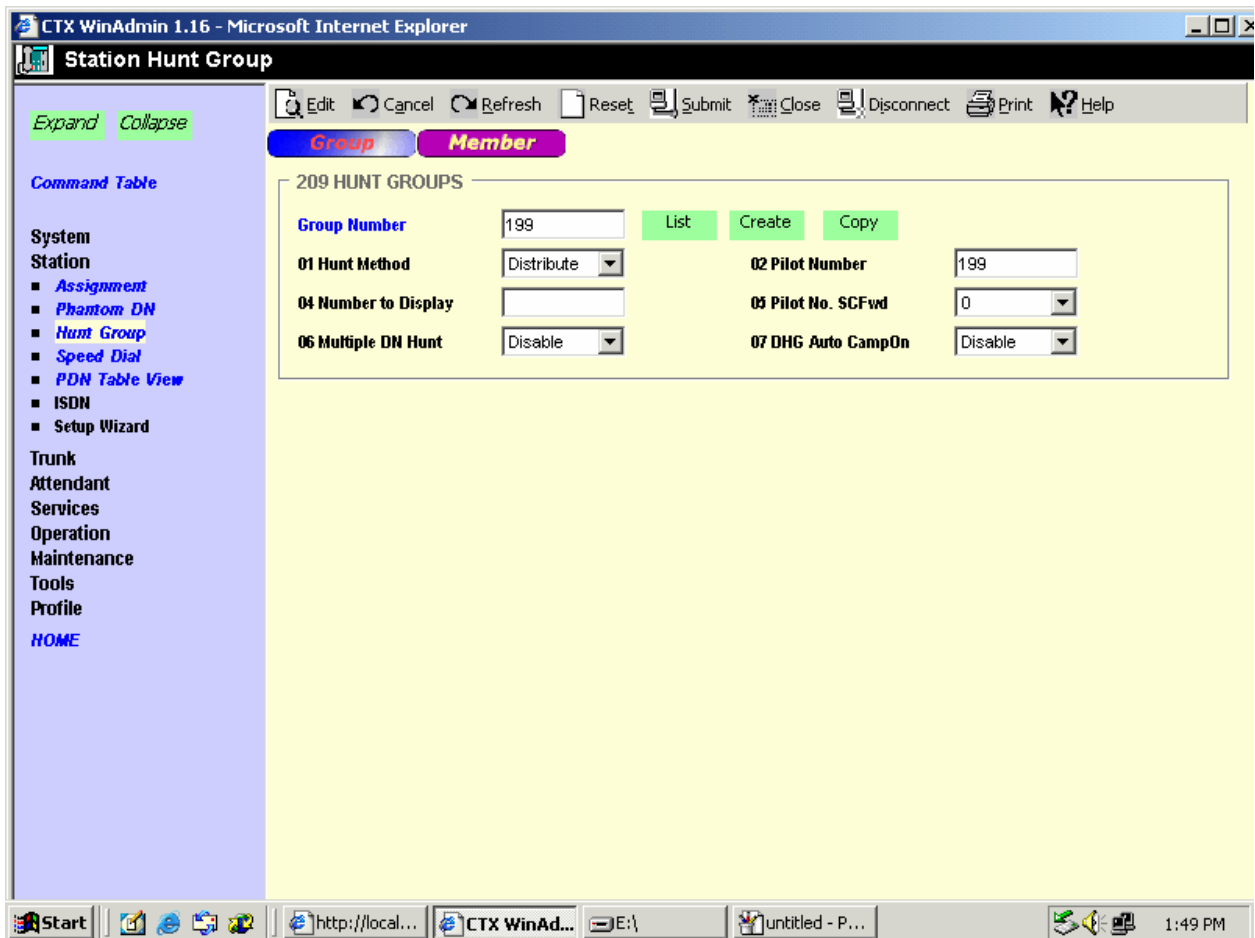
Important notes about the above programming:

1. The `Control Method` field should be set to `Inband/DTMF`.
2. The `Send A/D Tone` field should be set to `Enabled`.
3. The `End-to-End` field should be set to `Enabled`.

6.3   Setting Up Hunt Group

Use the `Station Hunt Group` menu selection to define a hunt group that includes all the configured analog station ports you wish to use for the gateway. The example below shows performing this command.

Important notes about the above programming:

1. The `Group Number` defines the hunt group index number in the PBX system.
2. The `Pilot Number` field defines the extension provided to the hunt group and should be used as the forwarding point for all subscriber stations as well as the inbound call point o access the server.
3. The `Hunt Method` field can be adjusted as required to change between various hunting patterns.

Once the hunt group is defined you use the `Member` tab to add all your analog voice mail ports to the group. The example below shows performing this command.
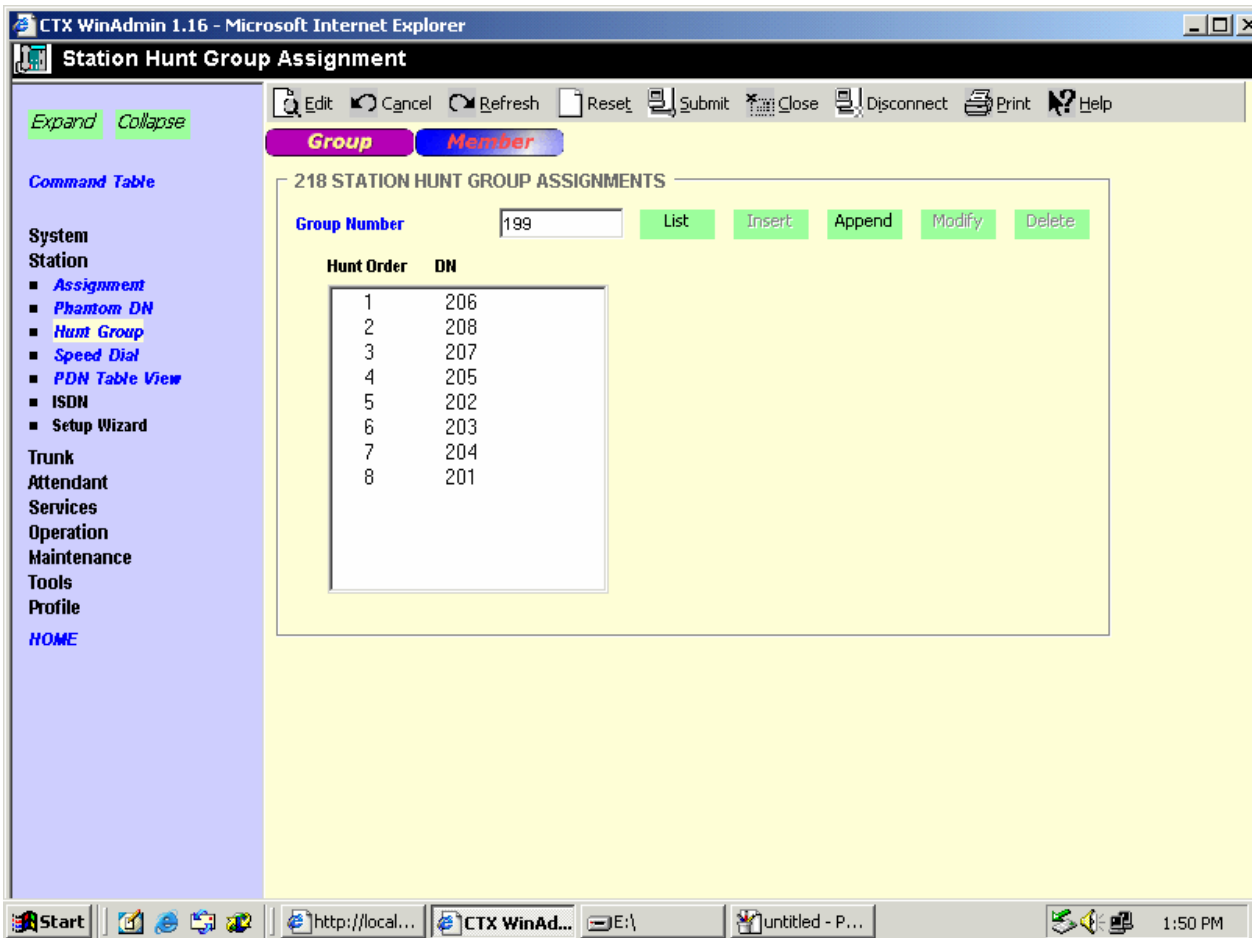
NOTE: The order that the ports appear on this page (the order that the ports are included in the hunt group) needs to match the order that the physical ports are plugged into the gateway(s) as well as the order that the extension numbers are configured into the LTN fields on the gateway(s). The there is a miss-alignment between the PBX configuration, the gateway configuration and or the physical connections then the data will not be properly aligned with the call when it arrives to the gateway. This can result in either a lack of integration or cross-integrated calls.

6.4   Setting Up Subscriber Station Sets

All the forwarding of the subscriber station sets can be defined directly on subscriber station set using either feature access codes or the phones soft menu keys. Subscribers 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.

6.5   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 not be shown as it is not provided by the PBX. 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 will not be shown as it is not provided by the PBX. 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 not be shown as it is not provided by the PBX. 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 not be shown as it is not provided by the PBX. 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 not be shown as it is not provided by the PBX. 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 not be shown as it is not provided by the PBX. 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 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 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 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 subscribers light on that is currently off.	This should return success.
20	Turn a subscribers light on that is currently on.	This should return success.
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 capture 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 server.
- `tel event` and `tel code` – This allows the capture 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:
    - o Call control problems (dropped calls, failing transfers, etc…)
    - o 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 capture of tone related audio 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:
    - o Failing transfers
    - o Failing outbound calls (play to phone)
    - o Dropped calls (callers cut off while leaving messages, 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.

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

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