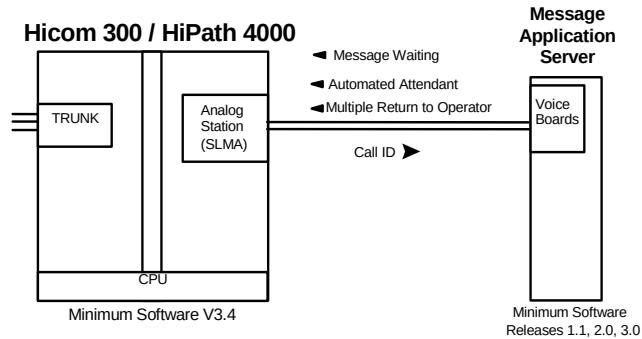


Configuration Note 88526 – Version H (1/10)

Siemens Hicom 300 /300 E, HiPath 4000 (EMEA)



**With Inband integration,
one pathway between the
PBX and the Avaya
Message Application Server
transmits both call
information and voice
communications**

MAS Requirements

¹Release Note:

Should features of the integration not function optimally when integrated to a PBX or MM that may be operating on an unsupported software release as defined Section 2.0 and 3.1, customers will need to upgrade their PBX and/or MM to a supported software release.

PBX hardware requirements

1.0 METHOD OF INTEGRATION

With Inband integration, one pathway between the PBX and the Avaya Message Application Server (MAS) transmits both call information and voice communications. The pathway is provided by 2-wire analog single-line circuits that connect to Dialogic cards in the MAS. Each Dialogic port simulates 2-wire analog lines. Calls to a MAS port are preceded by the called party information from the PBX in DTMF format. The MAS then answers and plays the appropriate greeting. Message Waiting Indications (MWI) are set and canceled by dialing a feature access code followed by the extension number.

2.0 AVAYA MESSAGE APPLICATION SERVER REQUIREMENTS

- Dialogic D/41 or D/120 cards (4 and 12 ports/cards)
- Supported Releases ¹: 1.1, 2.0, 3.0, 3.1, 4.0, 5.x

3.0 PBX HARDWARE REQUIREMENTS

- Analog 2500 stations per port - (SLMA-24 (Q2246-x))

NOTE: The customer must provide the necessary hardware. Also, verify the system has enough DTMF registers for Inband applications.

- Cables:
 - RJ11 four-wire telephone cord, one per MAS port

PBX software requirements

3.1 PBX SOFTWARE REQUIREMENTS

- Software [1 \(see page 1\)](#): V3.4 or above

Supported integration features

4.0 SUPPORTED INTEGRATION FEATURES

[✓] Items are supported

System Forward to Personal Greeting

All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]
Busy/No Answer	[]

Station Forward to Personal Greeting

All Calls	[]
Ring/no answer	[]
Busy	[]
Busy/No Answer	[]

Auto Attendant	[✓]
Call Me	[✓]
Direct Call	[✓]
External Call ID (ANI)	[]
Find Me	[]
Internal Call ID	[✓]
Message Waiting	[✓]
Multiple Call Forward (refer to section 8.6)	[✓]
Multiple Greetings	[]
Multiple Return to Operator	[✓]
N+1	[]
Outcalling	[✓]

IMPORTANT: PBX options or features not described in this Configuration Note are not supported with this integration. To implement options/features not described in this document, please contact the Avaya Switch Integration product manager.

Configuring the Siemens PBX to integrate with the MAS

5.0 CONFIGURING THE SIEMENS PBX TO INTEGRATE

Before you begin programming, it is recommended that a hard copy of the customer database be obtained to verify existing programming. The following programming is intended for certified PBX technicians/engineers.

Not all administrative possibilities are shown. The scope of this document does not include discussing all features and their interactions. Initial Startup or Default values on the Switch are assumed.

- ☐ Set the system parameters in the ZAND
- ☐ Create the Analog voice ports
- ☐ Create the system access Hunt Group
- ☐ Configure the ZIEL for the call type string
- ☐ Set the display name for the MAS ports
- ☐ Set the rights for the MAS ports to set/cancel MWI

Set system parameters

5.1 SET THE SYSTEM PARAMETERS

Allow all other fields to default.

```
Zand: Datenall,
      PREDIA=JA (Enables DTMF predialling)
      NAWAVERZ=JA (creates a pause between offhook of the voicemail system and actually sending the DTMF integration string)
```

```
CHANGE-ZAND:TYPE=ALLDATA,PREDIA=YES,POSTDDLY=YES;
```

NOTE: The PBX must have the Internal Dial Tone parameter set to HIGH frequency, not LOW frequency or stutter dial tone. Otherwise MM (Dialogic) will not recognize dial tone.

Programming analog voice ports

5.2 CREATE THE ANALOG VOICE PORTS

Create the analog voice ports. Below is an example for the programming:

```
einr-scsu: tlnnu=2400, lage=1-4-109-0, gerkonata,
           waba=0, vbz=10, cos1=0, cos2=1, rngi=nein,
           cofidx=0, sond=nawa&sotl,
           dpar=mfvtn&predia&asend&bsend, wahl=mfv;
```

Please ensure they contain the following attributes:

```
sond=nawa&sotl
```

```
dhpar=mfv tln&predia&asend&bsend
wahl=mfv
```

ADD-

```
CSU:STNO=2400,DVCFIG=ANATE,DPLN=0,ITR=0,COS1=1,COS2=1,LCOSV1=
1,LCOSV2=1,LCOSD1=1,LCOSD2=1,SPEC=SSTN&SUFDIAL,DIAL=DTMF,DHP
AR=ASEND&BSEND&PRE DIA&DTMFST;
```

Create the system access hunt group**5.3 CREATE THE SYSTEM ACCESS HUNT GROUP**

Create the main System Access hunt group and assign the analog voice port extensions as members. This hunt group will be the forwarding target for all subscribers.

Note: The hunt group number is not a separate number, but a physical port number on the UM system. Normally port 1 is used.

Below an example for a 4-port linear hunt group 2400 with hunt members 2401 to 2403:

```
einr-sa:spr, 2400,,, 2401&2402&2403, ltn, 0, MM
messages1,,ja;
```

ADD-

```
SA:TYPE=VCE,CD=2400,STNO=2401&2402&2403,STYPE=LIN,CQMAX=0,NA
ME="MM MESSAGES1",VARCQ=Y;
```

Note: Ensure all voicemail subscribers are system or station forwarded to the voicemail hunt group, or DTMF call handling will not be passed.

Configure the DTMF call type strings**5.4 CONFIGURE THE 'ZIEL' FOR THE CALL TYPES**

```
einr-ziel:vmx,dri,2400,***1; (direct internal call)
einr-ziel:vmx,dra,2400,***2; (direct external call)
einr-ziel:vmx,aui,2400,***3; (forward all calls)
einr-ziel:vmx,rws,2400,***4; (forward on RNA or busy)
```

```
ADD-ZIEL:TYPE=EVMTAB,SZTYPE=DSZINT,EVMCD=2400,SZCD=***1;
```

```
ADD-ZIEL:TYPE=EVMTAB,SZTYPE=DSZCO,EVMCD=2400,SZCD=***2;
```

```
ADD-ZIEL:TYPE=EVMTAB,SZTYPE=FWD,EVMCD=2400,SZCD=***3;
```

```
ADD-ZIEL:TYPE=EVMTAB,SZTYPE=FWDNANS,EVMCD=2400,SZCD=***4;
```

Note: The call types are programmed on the hunt group of the MM system. In this example 2400 is the main hunt group number.

5.5 SET THE DISPLAY NAME FOR THE MM SYSTEM

```
einr-persi:tln, 2400, Modular Messaging;
```

Set the name that appears in the display of a caller if the MM hunt group is called internally

ADD-PERSI:TYPE=STN,STNO=2400,NAME="MODULAR MESSAGING";

Note: 2400 is the number of the MM hunt group. This name will appear on the display if a user calls the MM hunt group internally

Give the MM ports the rights to set/cancel MWI of user stations

5.6 SET RIGHTS FOR MM PORTS TO SET/CANCEL MWI

This is an example for the configuration of a port extension for MWI. Repeat this step for every port you wish to allow for MWI. It is recommended that you enable all MM ports for it.

einr-ssc:1-4-109-0,mmmwi,0,0,1,0,ja;

ADD-SSC:PEN=1 – 4 – 109 – 0,TYPE=VMXMWI,MSTO=0,COFIDX=0,COS=1,LCOSSV=1,DPLN=0,INS=Y;

Message waiting indicators are set and canceled by dialing a feature access code followed by the extension number.

Message Waiting Parameters

Permit Message Waiting Lights	✓
Message Waiting Light Prefix ON	#*1
Message Waiting Light Prefix OFF	#*8

The DTMF codes for turning MWI on or off may vary from site to site. Please make sure you check with the PBX engineer which codes are actually set at your site!!!

The HiCom 300 does not require the same port to turn message-waiting light on or off.

Please refer to the Consideration section at the end of this document for special PBX programming considerations.

Configuring the MAS

6.0 CONFIGURING THE MESSAGE APPLICATION SERVER

Configuring the MAS platform for proper PBX integration requires configuring several folders accessed within the **Voice Mail System Configuration** application, and a certified MM engineer.

- Access the **Voice Mail System Configuration** application from the MAS program group.
- Access the **Voice Mail Domain**, and select **PBXs – Siemens Hicom**
- Access the **General** tab and set the following values:
 - **Go Off Hook when Port Disabled = Enable by checking the box**
 - Pause before Digits (ms) = 500**
 - Pause Interval for Comma in Dial String (ms) = 1000**
 - DTMF Inter-Digit Delay during Dialing (ms) = 80**
 - DTMF Length during Dialing (ms) = 80**
 - DTMF Inter-Digit Delay during Detection = 50**
 - DTMF Length during Detection (ms) = 50**
- Access the **Call Transfer** tab and set the following values:
 - **Transfer Mode = Blind**
 - Transfer Prefix Code = &,XP**
 - Transfer Complete Code = Leave Blank**
 - Transfer Release Code when Busy = &,**
 - Transfer Release Code when No Answer = &,**
 - Transfer Release Code when Reject = &,**
 - Flash Time Interval (ms) = 80**
 - Enable Call Progress = Leave Blank**
- Access the **Hangup Detection** tab and set the following values:
 - **Maximum Continuous Tone before Hanging Up (ms) = 6000**
 - Hangup String = Leave Blank**
 - Hangup String Timeout (ms) = 0**
 - Minimum Duration For Drop in Loop Current (ms) = 300**
 - Maximum Silence before Hanging Up (ms) = 6000**
- Return to the **Voice Mail System Configuration** window, and select **Voice Servers** and access **Telephony Interface (Analog)**.
- Select the **Analog** tab to configure the selected port(s) on your Voice Server as follows:
 - **Playback Volume = 2 (Default)**
 - **Number of Ports = Enter the number of ports in your system**

Note: The MAS service must be restarted to allow port enabling

- Enable the port(s) by checking the **Box** field next to the Port field
- **Extension** = Enter the proper extension number assigned to each port
- **Incoming Ring Count** = 1
- **Primary ID** = Leave Blank
- **Secondary ID** = Leave Blank

- Return to the **Voice Servers** section and access **PBX Integration**.

- Access the **General** tab. Select **Inband** as the **Integration Type**.
- Access the **Inband** tab and set the following values:

- **Maximum Inter-digit Gap (ms)** = 500
Pause before Inband Digits (ms) = 1500
DTMF On Time (ms) = 80
Search Entire String for Reason Code = Leave Blank
Location of Inband reason code = 1
Log Inband Packets = Leave Blank
Fixed Length Packets = Enable by checking the box
Filler Character = *
Delimiter Character = <None>
Right Alignment of Digits in a Field = Leave Blank
Request String Supported = Leave Blank

- Access the **Protocol Settings** button within the same tab and set the following values:

Show Advance Call Packet Type = Leave Blank

Call Packet Type = Divert:

Codes for Call Type:

Code 1 = ***4

Code 2 = ***3

Field Type Settings:

Called Id = 11 **Length** = 13

Calling Id = 5 **Length** = 3

Trunk Id = Leave Blank.

Call Packet Type = Direct:

Codes for Call Type:

Code 1 = ***1

Field Type Settings:

Called Id = Leave Blank

Calling Id = 5 Length = 7
Trunk Id = Leave Blank.

Note: The Hicom 300 doesn't distinguish between call forwarding on RNA or busy, therefore the divert type is used for these calls.

Note: The fixed length of the number integration fields can be programmed in the PBX between 6 and 15 digits. Please check the integration string sent by the switch with the Operational History Trace program and change the 'Length' settings accordingly.

For example, on a Direct Call Packet Type using 3 digit extensions (e.g. 899), you would use:

Calling ID = 5 Length = 7

In this example, the History Trace should indicate: *1899, with ***1 as the Direct code followed by 899, the extension number with a total of 7 digits.**

- Return to the **Inband** tab and access the **MWI Settings** button within the same tab and set the following values:

Port Group Name = Select the port group to be used for MWI

Max. MWI Sessions = Enter the maximum number of MWI sessions allowed at one time. The Default value is 1.

Indicator On/Off signals must use same port = Leave Blank

Trunk Id = Leave Blank

Indicator On: Prefix = #*1 Suffix = Leave Blank

Indicator Off: Prefix = #*8 Suffix = Leave Blank

Note: Make sure you have checked with the PBX engineer that your MWI entries match with the programming of the Hicom.

After making these changes, return to "Configuring the voicemail system" within the Message Application Server Installation guide. Ensure you are prompted to restart the Message Application Server services to apply these changes.

8.0 CONSIDERATIONS/ALTERNATIVES

8.1 To be able to test the integration it is essential to have the tracing server running and to check any incoming integration string digits on a live trace.

Important notes regarding this integration

Additionally, it may help to have an analog phone/device present for testing, that displays received DTMF digits thus being able to locate a problem in case the MAS is not receiving all the digits of an integration string or not recognizing digits properly.

8.2 The PBX does not allow a call to be transferred unsupervised to an extension that is busy and not busy forwarded. This is to prevent a caller from being transferred to a busy signal. Instead, the call will be sent back to the extension of the person attempting to transfer the call. This affects return-to-operator and unsupervised automated attendant transfers. Any station that can be reached through such applications must have busy forwarding targets programmed. Otherwise, callers might be returned to MAS when attempting to transfer out of it.

8.3 The 'busy' sent as a disconnect signal may vary. Depending on the site and whether the call originated internally or externally (e.g. mobile phone) the disconnect busy signal may vary. Please ensure you test the disconnect recognition with internal, external, mobile phone, and if applicable, internal calls.

8.4 First digit or digits of the integration string are not received/shown by the MAS. If this happens, please have the Hicom engineer change the timer for the NAWAVERZ parameter in the DTIM2 and confirm NAWAVERZ is set to YES in the ZAND.

- Hicom command: AE-DTIM2:ALLG,,,,,,,600;
- MAS: Lower the parameter 'pause before digits' on the PBX/general tab.

If only the 1st digit is missing, then alternatively you could add the call types without the first digit (i.e. **1 instead of ***1) as call type codes.

8.5 System doesn't put call on hold and caller hears MAS dial DTMF digits. This means that the 'Flash time' on the 'Call Transfer' tab of the PBX settings is not correct for your site. You may have to experiment with settings between 60ms and 110ms. If the caller gets disconnected instead of being put on hold it means that the flash time is too long.

8.6 Personal greeting of original-called party on double-call forward, is only supported with the **Hicom 300 E Version 6.4** and above.

8.7 Call is not forwarded on double forward. This happens if one station is using call forward (busy/RNA) and one station is set to forward all calls. For these are 2 different features for the Hicom it doesn't allow for mixing them. It normally results in the call 'hanging' at the last station and not being forwarded to MAS.

8.8 CorNet is an Inter-PBX ISDN Signaling System that provides feature transparency across a PBX network. All locally supported integration features are available over CorNet (when operating within a Hicom 300 E environment). In addition:

- There are no additional MAS requirements necessary in order to work with CorNet.
- Mailboxes that have extensions on remote PBXs require those extensions to be in the Local PBX DialPLN.
- The remote extensions will require MB in their Class of Service (as do local extensions).
- Remote extensions will need to forward calls to the MAS hunt pilot number (as do local extensions).

8.9 DTMF inband signaling.

Section	Length	Format
Call Type	fixed length of 4 digits	***n (n=1,2,3,4)
CLI A-TLN	Fixed length between 6 and 16 digits	Number left aligned Filling character = * (to fill the field up to the fixed length) e.g. nnn*** = extension number with 3 digits 000000 = blocked/secret number 0111111 = number known, but length greater than field length 022222 = number unknown.
CPI B-TLN	Fixed length between 6 and 16 digits. Also variable length is possible.	Number will only be sent on forwarded calls. Number left aligned. If field length is fixed, the settings of CLI will also apply.

CHANGE HISTORY		
Revision	Issue Date	Reason for Change
Version A	04/28/04	GA
Version B	09/16/04	Updated to meet MM R2.0
Version C	09/12/05	Updated to support the HiPath 4000
Version D	04/11/06	Updated to meet MM R3.0
Version E	05/05/08	Updated to support MM 4.0
Version F	2/02/09	Updated to support MM 5.0
Version G	07/09	Updated to support MM 5.1
Version H	01/19/10	Added note indicator to titles of Section 2.0 and 3.1; added corresponding note in sidebar; removed word “supported” in same sections for MAS releases and PBX Software releases

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AVAYA Inc.
1033 McCarthy Blvd.
Milpitas, CA 95035
(408) 577- 7000
<http://www.avaya.com>