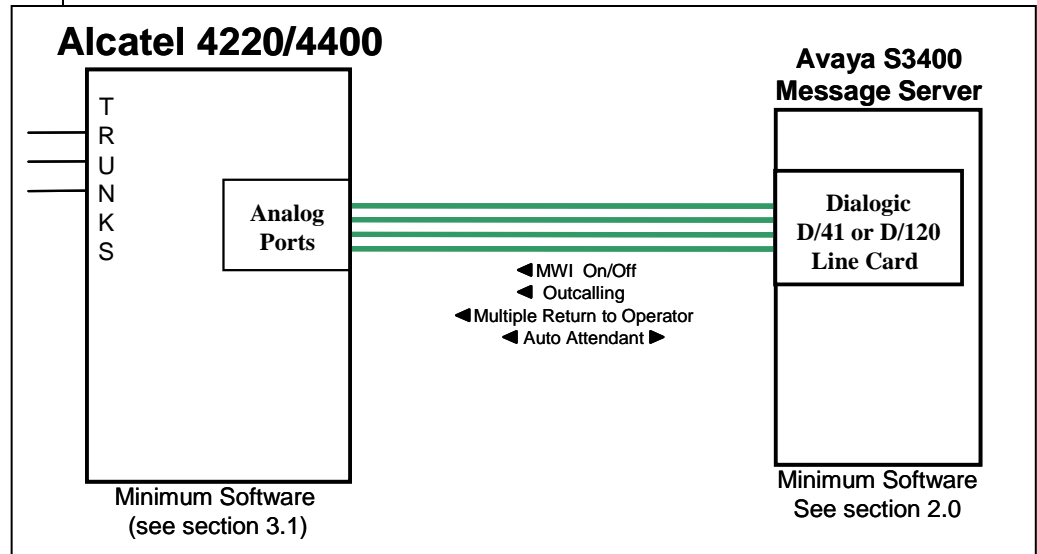


Alcatel 4220/4400 - (EMEA)



With Inband integration, one pathway between the PBX and the 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.

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 card in the MAS. Each Dialogic port simulates 2-wire analog lines. Calls to an 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 Indicators are set and canceled by dialing a feature access code.

2.0 AVAYA MESSAGE APPLICATION SERVER REQUIREMENTS

- **Dialogic D/41 or D/120 cards** (4 and 12 ports/cards)
- **Software Releases ¹:** 1.1, 2.0, 3.x, 4.0, 5.x

PBX hardware requirements

3.0 PBX HARDWARE REQUIREMENTS

- Station appearances, one per MAS port
Linecard Z12 – 12 ports
Linecard Z24 – 24 ports
- **Cables:**
- RJ11 four-wire telephone cord, one per MAS port

NOTE: The customer must provide the necessary hardware.

PBX software requirements

3.1 PBX SOFTWARE REQUIREMENTS

- Minimum Software 4220 ^{1 (see page 1)}: R1.3
R1.4
R1.5
R1.53
- Minimum Software 4400 ^{1 (see page 1)}: R5.0.1*
- VPS Licensing: Each voice processing (non-MWI) port requires a VPS license.

***Note:** *This was the Alcatel 4400 software release used for validation.*

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	[]
Multiple Greetings	[✓]
Multiple Return to Operator	[✓]
N+1	[✓]
Outcalling	[✓]

Configuring the Alcatel to integrate with the MAS

5.0 CONFIGURING THE ALCATEL TO INTEGRATE

- The Alcatel 4220/4400 PBX configuration is based on a Windows GUI.
- Final check can be done using a text mode screen: see tables later in the document.
- No GUI screen capture is provided.
- Main steps are listed for Alcatel FE
- Check that feature bit "VPS protocol" is available (VPS protocol Alcatel's proprietary DTMF Inband integration protocol for external message servers)
- Check the authorized number of voicemail ports equal the number of ports on the message server
- Configure each VPS ports with a directory number, address, and **type 4620 (VPS with CLIP)**. **NOTE: The parameters below should be set as follows:**

Parameter	Setting
DTMF for VPS Protocol	NO for Alcatel 4400 YES for Alcatel 4200 * (see below)
Tone for VPS Protocol	YES

* On the **4200** it is possible to change the DTMF code for Dial Tone from B5 to B, but on the Alcatel OmniPCX Enterprise (4400) this is not possible. This affects Message Waiting Indicator and hang-up detection. See the considerations section at the end of this configuration note.

- Configure the voicemail hunt group with all VPS ports
- Check that the directory number of VPS ports and the hunt group number is displayed in the screen "APPLICATION / NUMERO ANNUAIRE MESSAGERIE VOCALE"
- Configure mailbox owners with the hunt group number in "NUMERO DE MESSAGERIE" field and check the parameter "ENVOI DU CODE D'ACCESS"
- Configure analog extension owners with parameter "LAMPE MESSAGE" = OUI
- Configure a access code for the feature "CONSULTATION DE LA MESSAGERIE VOCALE"
Internal access code for quick logon: 2 digits access code;
calling the internal hunt group number generate an external call: no quick logon available
- Configure MWI ON access code using parameter "CONSULTER MESSAGE"
- Configure a MWI OFF access code using the parameter "ANNULATION LAMPE MESSAGE"

- For digital set owners, allow access to the "direct access to personal greeting" feature, by configuring the parameter "dépôt des messages vocaux" in activated mode.
- For analog set owners, configure an access code for the "direct access to personal greeting" feature, using the parameter, "dépôt des messages vocaux" in the access code liste ("suffixe" list).

5.1 CENTRALISED VOICEMAIL

- Configuration of remote A4220 PBX node:
- Automatic propagation over the network configures all voicemail parameters in "APPLICATION" et "NUMERO ANNUAIRE MESSAGERIE VOCALE" screens for all remote nodes.
- In the "préfixe" list, configure the "TYPE MESSAGERIE VOCALE" parameter with the network access code of the remote centralized voicemail system.
- Configure mailbox owners with the network access code of the remote centralized voicemail system in the " NUMERO DE MESSAGERIE" parameter.
- Configure analog extension owners with parameter "LAMPE MESSAGE" = OUI

5.2 FINAL CHECK TEXT MODE SCREEN

- Voicemail hunt group:

◇ **Example uses: 3320, 4 ports : 3601 through 3604**

Groupes	1
No annuaire	3320
Nom d'annuaire	MESSAGERIE
Type Recherche	Sequentiel
% attentes autorise	0
No Categorie de connexion	0
Categorie acces res.public	4
No affectes au groupe	`3601`3602`3603`3604`
No Entite	2

Authorized features

Categories	1
No Categories d'expl.tel.	0

Exploitations locales\Appel rapide de l'associe	1
Exploitations locales\Consult.personnes a rappeler	1
Exploitations locales\Rappel dernier appelant local	1
Exploitations locales\Reponse recherche personne	0
Exploitations locales\Consultation mess.vocal	1
Exploitations locales\Rappel rendez-vous/Reveil	1
Exploitations locales\Test des tonalites	1
Exploitations locales\Aller retirer telex	0
Exploitations locales\Aller retirer texte	0
Exploitations locales\Aller retirer telecopie	0
Exploitations locales\Consulter messages	1
Exploitations locales\Consulter texte	0
Exploitations locales\Consulter image	0
Exploitations locales\Prefixes ACD	1
Exploitations locales\Conference programmee	0
Exploitations locales\Annul.Rendez-vous/Reveil	1
Exploitations locales\Annulation voyant message	1
Exploitations locales\Changement etats chambres	0
Exploitations locales\Mini bar	0
Exploitations locales\Acces gest.mess.vocale	0
Exploitations locales\Enreg.conversation	0
Exploitations locales\Adresse PABX en DPNSS	0
Exploitations locales\Appel direct rech.personne	0
Exploitations locales\Infocenter	0
Exploitations locales\Depot mess.vocale	1
Exploitations locales>Select.Ligne principale	0
Exploitations locales>Select.lignes secondaires	0
Exploitations locales\Numerotation Z derriere UA	0
Exploitations locales\Masque identite du correspondant	0
Renvoi mess.vocal	Appel messagerie p.final (*)

NB * : Multiple forwarding: Not allowed / CPI of first extension / CPI of last extension.

PBX numbering plan for voicemail

Traducteur	Numer	Exploitation en local
	0	
1	*	
1	0	

<i>I</i>	<i>1500</i>	<i>Consultation mess.vocal</i>
1	3300	
1	40	
1	41	
1	42	
1	43	Appel rapide de l'associe
1	44	Rappel dernier appelant local
1	45	
1	47	Consult.personnes a rappeler
1	48	
1	50	
1	51	Rappel rendez-vous/Reveil
1	52	Annul.Rendez-vous/Reveil
1	53	
1	54	
1	55	
1	56	
1	57	
1	58	
1	59	
1	60	
1	61	
1	62	
1	63	
1	64	
1	65	
1	67	
<i>I</i>	<i>68</i>	<i>Consulter messages</i>
<i>I</i>	<i>69</i>	<i>Annulation voyant message</i>
1	70	
1	71	
1	72	
1	73	
1	74	
1	75	
1	76	
1	77	
1	78	
1	80	
1	81	
1	82	
1	83	

1	84	
1	85	
1	86	
1	87	
1	88	
1	89	
1	9	

User configuration

No annuaire	3330
Type de poste	4012
No Entite	2
<i>No poste associe</i>	3320
Categorie acces res.public	4
No Categories d'expl.tel.	0
<i>No annuaire Mess.Vocale</i>	3320
<i>Type Messagerie Vocale</i>	4620 (VPS)
Caracteristiques multilignes\Prise de ligne arrivee auto	Oui
Caracteristiques multilignes\Prise de ligne depart auto	Oui
Caracteristiques multilignes\filtrage selectif	Non
Caracteristiques multilignes\Debordement sur non reponse	Oui
Caracteristiques multilignes\Debordement sur occupation	Non

5.3 PROGRAMMING FORWARD TO PERSONAL GREETING

- **Immediate or All Calls**, will forward all calls.

Activation Code **60**

- **Busy Diversion**, will divert calls to a specified extension when the user is on the phone.

Activation Code **61**

- **Ring-no-Answer**, will divert a call if the user does not answer their phone within a specified time.

Activation Code **62**

- **Ring-no-Answer / Busy**, will divert a call if the user does not answer their phone within a specified time or if their are engaged on another call.

Activation Code **63**

Programming forward to personal greeting

- **Cancel ALL Diverts 64**

Note: PBX features codes are customer dependent.

6.0 CONFIGURING THE MESSAGE APPLICATION SERVER

Configuring the MAS

Configuring the MAS platform for proper PBX integration requires configuring several folders accessed within the **Voice Mail System Configuration** application.

- Access the **Voice Mail System Configuration** application from the MAS program group.
- Access the **Voice Mail Domain**, and select **PBXs: DHS2 (Alcatel 4220)**
- Access the **General** folder and set the following values:
 - **Go Off Hook when Port Disabled = Enable by checking the box**
 - **Pause before Digits (ms) = 550**
 - **Pause Interval for Comma in Dial String (ms) = 2000**
 - **DTMF Inter-Digit Delay during Dialing (ms) = 80**
 - **DTMF Length during Dialing (ms) = 80**
 - **DTMF Inter-Digit Delay during Detection = 50**
- Access the **Call Transfer** tab and set the following values:
 - **Transfer Mode = Blind**
 - **Transfer Prefix Code = &N***
 - **Transfer Complete Code = C**
 - **Transfer Release Code when Busy = &**
 - **Transfer Release Code when No Answer = &**
 - **Transfer Release Code when Reject = &**
 - **Flash Time Interval (ms) = 270**
 - **Enable Call Progress = Enable by checking the box**
 - **Start Delay for Call Progress (ms) = 1500**

***Note:** When integrating with an Alcatel 4400 we found a comma was needed for off PBX transfers. This would be inserted between the & and the N, the correct syntax then being: &,N.

- Access the **Hangup Detection** folder and set the following values:
 - **Maximum Continuous Tone before Hanging Up (ms) = 6000**
 - **Hangup String = B9* (see below)**
 - **Hangup String Timeout (ms) = 0**

Minimum Duration For Drop in Loop Current (ms) = 300
Maximum Silence before Hanging Up (ms) = 6000

- * With the **Alcatel 4200** the DTMF for VPS Protocol is set to YES; this means that the PBX still send codes for disconnect (B9), Dial tone etc.
With the Alcatel 4400 we must switch DTMF for VPS Protocol to NO; This means that the PBX will not send a disconnect tone. See the special considerations at the end of this note for more information.

Support NOTE
for MM 1.1 & 2.0 only:

- Limit Requests should be checked.
- Maximum Requests per Minute should be set for 200.

- Access the **Message Waiting Indicator (MWI)** tab
Enable Message Waiting Indicator (MWI) = Enable by checking the box
MAS MWI Server = Enter the name of the MWI server created during the installation procedure.
Limit Requests = Leave Unchecked
Maximum Requests per Minute = <grayed out>
Message Application Servers that Support MWI = This box should contain a list of MAS servers capable of placing MWI requests.

- Next select **Voice Servers** and access **Telephony Interface (Dialogic Analog)**
Select the **Analog** tab
Playback Volume = 2
Number of Ports = Enter the number of ports in your system
Note: The MAS service must be restarted to allow port enabling

Enable each **Port** by checking the checkbox 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

- Next access the **General** folder. Select **Inband** as the **Integration Type**.
- Access the **Inband** property folder. Insure the following settings:
 - **Maximum Inter-digit Gap (ms) = 500**
 - **Pause before Inband Digits (ms) = 3000**
 - **DTMF On Time (ms) = 80**
 - **Search Entire String for Reason Code = Leave Blank**
 - **Location of Inband reason code = 1**
 - **Log Inband Packets = Enable by checking the box**
 - **Fixed Length Packets = NO**
 - **Filler Character = <None>**
 - **Delimiter Character = ***
 - **Right Alignment of Digits in a Field = Leave Blank**
 - **Request String Supported = Leave Blank**

- Access the **Protocol Settings** button within the same folder. Set the following parameters:

Note: To expose all call types tick the ‘Show Advanced Call Packet Type’ tick box. This exposes the Auto Logon call type.

Call Type: DIVERT

Packet Length: blank

Code 1: A1

Code 2: AA1

Code 3: blank

Code 4: blank

Field Type	Sequence Number
Called ID	1
Calling ID	2
Trunk ID	3

Call Type: No Answer Extension

Packet Length: blank

Code 1: A6

Code 2: AA6

Code 3: blank

Code 4: blank

Field Type	Sequence Number
Called ID	1
Calling ID	2
Trunk ID	3

Call Type: BUSY

Packet Length: blank

Code 1: A5

Code 2: AA5

Code 3: blank

Code 4: blank

Field Type	Sequence Number
Called ID	1
Calling ID	2
Trunk ID	3

Call Type: Auto Logon

Packet Length: blank

Code 1: A2

Code 2: blank

Code 3: blank

Code 4: blank

Field Type	Sequence Number
Called ID	2
Calling ID	1
Trunk ID	3

The call types Check MBX, Confirm MBX and Transfer Initiate are not used.

- 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 = 68 Suffix = Leave Blank

Indicator Off: Prefix = 69 Suffix = Leave Blank

Note: MWI settings may vary check settings with PBX engineer.

After making these changes, return to “Configuring the voicemail system” within the S3400 Message Server Installation guide. Ensure you are prompted to restart the Message Application Server services to apply these changes.

- continued next page -

Appendix A - CALL PROGRESS TONES

Important notes regarding this integration

Called extension available (ringing)	B6 + Silence
Called extension goes off hook	C
Called extension is busy	B7 + Busy Tone
Called extension is invalid	B8 + Fast Busy
Dial tone (off-hook / flash-hook)	B5
Caller hang-up	B9
Called number available (external)	B6
Called number goes of hook (external)	None
Called number is busy (external)	B9

NOTE: Use only "B" as the dial tone instead of "B5." This avoids glare on the MWI ports. Otherwise the MWI ports do not work properly.

This is only possible on the Alcatel 4200.

For the Alcatel 4400 the codes cannot be changed and this is the reason that the DTMF for VPS Protocol is set to NO on this platform. Because of this, the 4400 will not send call progress tones, which has implications for hang-up detection; without a B9 signal calls do not clear. For this reason it is vital that a working tone set file is created.

8.0 CONSIDERATIONS/ALTERNATIVES

Important notes regarding this integration

- 8.1 Transfers off-PBX may experience incomplete or returned calls. Placing a pause character after the station and before the transfer complete code may resolve this issue.**
- 8.2 VPS with CLIP (set type 4620) must be configured for the voice ports.** This allows the PBX to send the Called, Calling and Reason codes to the voicemail system. Without this configuration (if VPS without CLIP (set type 4610) is used) there will be no calling party information or reason code, therefore all messages will appear from the external caller so cannot be replied to, and call handling for message greetings will not work.
- Please ensure this is set properly as noted in Section 5.0 in this configuration note, or problems may be encountered with the integration.
- 8.3 Each MWI port should be configured as a VPS port, exactly like the voice ports. (see Addendum section on MWI on p.17)**

CHANGE HISTORY		
Revision	Issue Date	Reason for Change
Version A	06/17/04	GA
Version B	09/16/04	Updated to meet MM R2.0
Version C	11/15/05	Updated with 4400 information.
Version D	12/2/05	Updated changes: <ul style="list-style-type: none"> • DTMF for VPS set to NO • Tone for VPS set to YES • Changed codes in Section 6 for Call Packet Types and added Optional Auto Logon instruction.
Version E	1/13/06	Changes: <ul style="list-style-type: none"> • PBX Ports now set to VPS “on.” Alcatel tech discovered how to get it to work properly. • Removed notes in test to the contrary. • MM integration setting for Fixed Length Packets now set to “NO” • Moved VPS setting table found in Consideration 8.2 to section 3.0 and rewrote 8.2 to ensure VPS is set correctly. • Added ADDENDUM at end of CN.
Version F	3/29/06	<ul style="list-style-type: none"> • Change MWI information and removed older/incorrect MWI screen in Addendum • Added Consideration 8.3 - All ports, including MWI ports, should be setup as standard VPS voice processing ports.
Version G	04/11/06	Updated to meet MM 3.0
Version H	05/05/08	Updated to support MM 4.0
Version I	2/02/09	Updated to support MM 5.0; Limit Requests for MWI changed in Section 6.0
Version J	07/09	Updated to support MM 5.1
Version K	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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ADDENDUM FOR ALCATEL 4200 / 4400 - PORTS & INTEGRATION

PBX CONFIGURATION

Each voicemail port is configured as 4620 (VPS with CLIP). Switching CLIP on tells the PBX to send the Called, Calling and Reason codes to the voicemail; with CLIP switched off we do not get Calling party information and so all calls appear to be from External Caller.

Below is a screen shot of the configuration of one of the PBX ports, but the important thing is that it is set to 4620 VPS + CLIP.

Note: CLIP is a licensed option so the PBX vendor must ensure that the system has enough licenses to cover the amount of voice ports required.

The screenshot shows a configuration window titled "Edit \Kelloggs\Users:860". The window contains a list of configuration fields for a voicemail port. The "Set Type" field is set to "4620 (VPS + CLIP)". Other fields include "Directory Number" (860), "Directory name" (VOICE), "Directory First Name" (MAIL), "Location Node" (1), "Shelf Address" (0), "Board Address" (23), "Equipment Address" (12), "Entity Number" (1), "Set Function" (Default), "Profile Name" (empty), "Key Profile" (None), and "Identifier of Domain" (0). The window also has "Ok", "Previous", "Next", and "Close" buttons at the bottom. The Windows taskbar at the bottom shows the Start button and several open applications, including "N...", "CMIS...", "F...", "V...", "D...", and "E...". The system clock shows 13:34.

Field	Value
Directory Number	860
Directory name	VOICE
Directory First Name	MAIL
Location Node	1
Shelf Address	0
Board Address	23
Equipment Address	12
Set Type	4620 (VPS + CLIP)
Entity Number	1
Set Function	Default
Profile Name	
Key Profile	None
Identifier of Domain	0
Add On Module 1	
Add On Module 2	
Add On Module 3	
External Alphanumeric Keyboard	

VOICE MAIL SYSTEM CONFIGURATION

Switching on CLIP results in the system sending different integration packets for internal and external calls, e.g. for internal Ring No Answer calls to extension 7001 from 7002 the packet would be A67001*7002 where for external calls it would be AA67001*7002. For this reason we make the following changes to the PBX Integration object in Voice Mail System Configuration:

Object: Integration
 Page: General
 Option: Fixed Length Packets: NO
 Option: Delimiter Character: *

Protocol Settings

These are the protocol settings, adjusted for variable length packets.

Call Type: DIVERT
 Packet Length: blank
 Code 1: A1
 Code 2: AA1
 Code 3: blank
 Code 4: blank

Field Type	Sequence Number
Called ID	1
Calling ID	2
Trunk ID	3

For the other call types that are used we specify the same structure as above, these are the codes for each type:

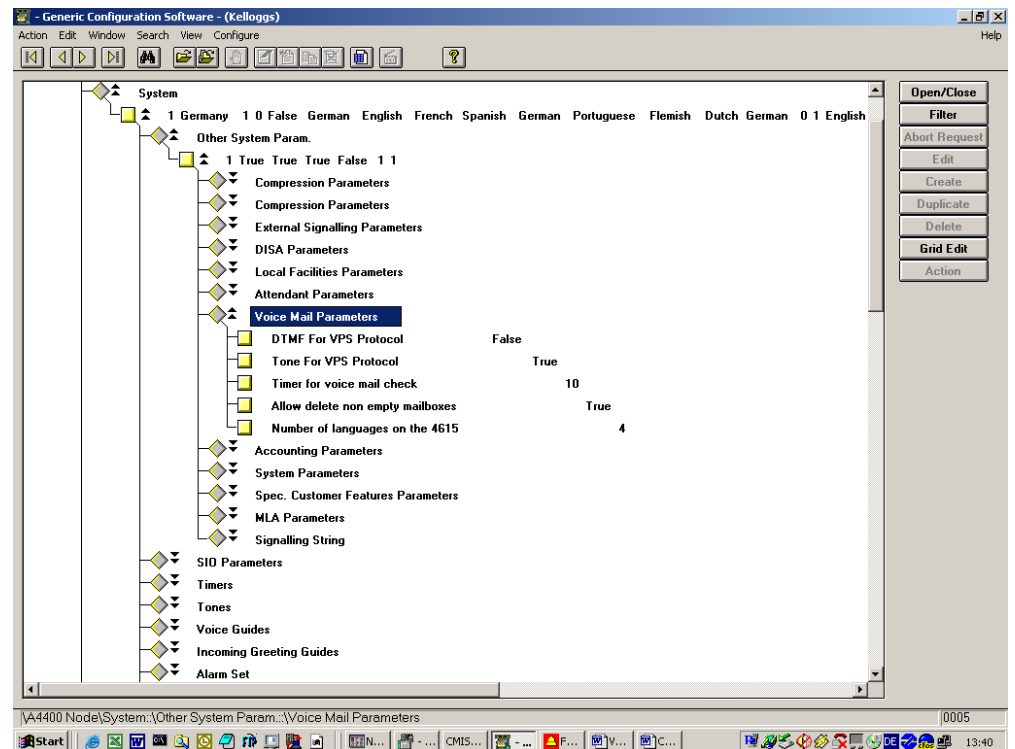
- No Answer Extension: A1 & AA1
- Busy Extension: A5 & AA5
- Direct Call: Not Used
- Immediate Transfer: Not used
- Auto Logon: A2 (Code 2 not used)
- Check MBX: Not Used
- Confirm MBX: Not Used
- Transfer Initiate: Not Used

VPS PROTOCOL GENERAL

The VPS protocol settings must be as follows:

- DTMF for VPS protocol: OFF
- Tone for VPS protocol: ON

Below is a screen shot of the VPS protocol settings:



Note: Switching off DTMF for VPS Protocol results in the PBX not sending a disconnect signal, which is specified in the revision C note. This means that without a working tone set file there is no hang-up detection, therefore it is vital that a working tone set file is used.

MESSAGE WAITING INDICATOR – *Operation & Caveats*

When a voicemail is left and the LED on the phone is switched ON, the Avaya MM actually sends a 'Message Deposit' signal to the PBX, which the PBX then relays to the phone set. In an Alcatel system 'Message' can mean voicemail, text, fax, or more importantly to us it is technically a recall request.

If the user presses the messages button and then chooses the option to access voicemail, they are technically answering the recall request that came from the MWI port. When they acknowledge the recall request they are calling the MWI port then being forwarded to the MM hunt group; this means that there is no auto logon. Instead the user hears the main greeting 'Thank you for calling....' and will then need to logon by pressing #. Once the recall request is acknowledged the MWI LED is turned off even though they have not yet logged on to listen to their voice message. (The MM would send an MWI OFF signal only after the voice message has been listened to.) If the user logged on and successfully listened to their messages, there is no problem; in the background the MM will send the MWI OFF signal which keeps the LED / mailbox in sync.

The potential for problem exists when a user presses the messages button, then for some reason does not complete the call into voicemail and listen to their message. Doing this will turn off the LED, but since the message was not listened to MM believes the LED remains on. Because of this the phone is now in an inconsistent state; there is an unread voicemail but no LED illuminated on the phone. Should a new message arrive in this user's mailbox, the MM will not send an MWI ON request because it believes that the LED is already on. Only when the user has logged on and listened to all voicemail will the MM send an MWI OFF signal thus returning the phone/mailbox to a consistent state.

However, should the user access the mailbox by dialing the pilot number, or pressing a short code such as ##, and then logging on (with the auto logon feature), the LED will not be turned off prematurely because they are not acknowledging the recall request. The lamp will only be switched off after message is played and the voicemail server sends an MWI OFF signal.

MWI Configuration & Correction

To avoid the potential issue explained above:

1. Configure ALL, including MWI ports, as standard VPS voice processing ports.
2. In the MAS you may optionally dedicate an MWI port group to ensure MWI will operate even when there is heavy call volume.

The message button on the phone will now properly recognize that the message is voicemail. Pressing the voicemail option will then take you to voicemail with auto logon, and the LED stays consistent.

MWI Port(s) must have an immediate diversion (forwarding) to the Voice-Mail huntgroup.

MWI is activated by the prefix 'collect text' followed by the extension. Deactivation is 'switch message LED off' followed by the extension.

In the voicemail system configuration, creating a separate port group for MWI is optional. This group may contain just this port (or more ports in a bigger system if required) to ensure you always have a port available for MWI. The MWI option in Voice Mail System Configuration is as follows:

- Port Group Name: *MWI*
- Max MWI Sessions: *Depends on how many ports are in the group.*
- Indicator On Prefix: *Depends on PBX configuration.*
- Indicator On Suffix: *blank.*
- Indicator Off Prefix: *Depends on PBX configuration.*
- Indicator Off Suffix: *blank.*
- Indicator On/Off signals must use the same port: *No. (do not ✓ box)*

ADDITIONAL VOICEMAIL HUNT GROUP INFO

Below is a screen shot of the voicemail hunt group configuration.

The screenshot shows the configuration window for a hunting group. The fields are as follows:

Directory Number	800
Directory Name	VOICE - MAIL
Identifier of Domain	0
Type of Hunting Group	RSVP Hunting Group
Subnetwork Reference	
Node Reference	
Search Type	Sequential
Group Busy With One Call	
Release After Timer	<input type="checkbox"/> True
Overflow Directory Number	
Authorized Camp on Calls	0
Connection Category Id	0
Public Network Category	2
Call Restriction Category	0
Withdrawal Authorized	<input type="checkbox"/> True
Dir. No Allocated to the group	01: 860
Entity Number	1

Note: For quick logon a prefix 'voicemail consultation is needed'.