

Release Note

Myriad and Halo series Deskphone

TC3097 ed.01 Release R140

Release Note FOR Alcatel-Lucent Enterprise Myriad and Halo series DeskPhone R140

This document provides the configuration details required to Alcatel-Lucent Enterprise Myriad and Halo series DeskPhone R140 connecting to 3rd party SIP Server.

Revision History

Edition 1: February 8, 2024 creation of the document

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1 Introduction

Firmware Version:

M3/M5/M7: 2.14.15.000.2490

M8: 2.14.15.000.2468

H3P/H3G/H6: 2.14.15.000.2484

You can download the firmware from https://www.aledevice.com/site/download.

2 New Features

- 1. Added support local conference manager.
- 2. Added support LDAP multi-line display support.
- 3. Added support M8 BT audio hub for PC.
- 4. Added support OpenVPN tar file import.
- 5. Added support separate wireless and wired parameters.
- 6. Added support time synchronization based on SIP signaling.
- 7. Added support new language Thai for mmi, Russian and Turkish for WEB.
- 8. Added support on-hook to transfer function.
- 9. Added support set ringtones for contacts.
- 10. Added support control to automatically generate a call park softkey on the call page.
- 11. Added support recording related functions, for file upload, automatic cleaning support.
- 12. Added support CDP.
- 13. Added support the contact template format can be csv.
- 14. Added support the different refuse code for different scenarios.
- 15. Added support port setting capability for pnp and ssh.
- 16. Added support custom tone.
- 17. Added support codec G726.
- 18. Added support call display source.
- 19. Added support customer language.



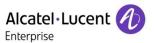
3 Optimization

3.1 Transfer optimization

Parameter	SIPTranSuccessfulNotify	config.xml
Description	It configures enable or disable the Semi-attend Transfer function.	
Permitted	false: disable feature	
Values	true: enable feature	
Default	false	
Parameter	SIPTranSuccessfulNotify	config.xml
Description	It configures after the Transfer operation is configured, which signaling is recindicates that the Transfer is successful, and the transfer process ends.	eived
	0 - NOTIFY containing a 2xx Status-Line	
	1 - NOTIFY containing a 100 Status-Line	
Permitted	2 - NOTIFY containing a 180 Status-Line	
Values	3 - REFER's 202 response	
Default	0	

3.2 BLF status display

Notify	Icon	LED State		Description
		M3/5/7	H3X6/M8	
Terminated	ನ	On (Blue)	On (Blue)	The monitoring account is idle. Note: If the notify message does not carry a clear status, it is regarded as idle.
Early/proceeding	.	Fast-flashing Blue	Fast-flashing Red	The monitoring account is ringing.
Confirmed	<i>®</i>	Slow-flashing Blue	On (Red)	The monitoring account is in call talking.
Confirmed-hold		On (Blue)	Slow-flashing Red	The monitoring account is on hold.
Parked	P	On (Blue)	Slow-flashing Red	The monitoring account is on parked.



Offline/Unregister		Off	Off	The monitoring account is offline or
	~			not registered.
	3			Not subscribed.
Unknow	A 94 94 94.0	Off	Off	The monitoring account is unknown,
	<u>ੂੰ</u>			that is, the Notify message carries
				other states than the preceding
				ones. or only receive 2000K, but no
				notify.

3.3 Audio delay optimization

Optimize voice creation delay for incoming calls, speed up voice creation time, and minimize audio loss during voice creation.

3.4 Aom & blf & blf list optimization

It is optimized to support a large number of BLF on AOM to improve stability.

3.5 Merge supports customer feedback and improves performance

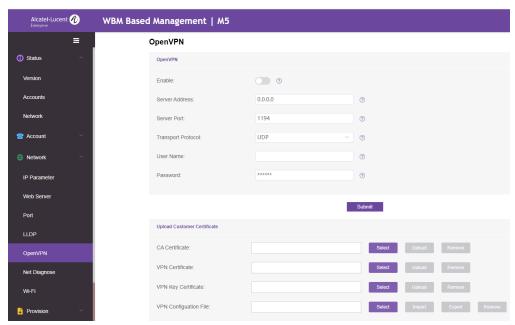
3.5.1 Optimized the OPENVPN deployment mode, added support for username authentication, and optimized the autop deployment mode for OpenVPN

The OpenVPN-related files include certificates (ca.crt and client.crt), key (client.key), userinfo(user.txt) and the configuration file (vpn.cnf) of the OpenVPN client.

The following table lists the unified directories of the OpenVPN certificates and key in the configuration file (vpn.cnf) for the ALE Myriad Series phones:

OpenVPN Files	Description	Unified Directories
ca.crt	CA certificate	/config/cert/openvpn/openvpn-ca.crt
client.crt	Client certificate	/config/cert/openvpn/openvpn-client.crt
client.key	Private key of the client	/config/cert/openvpn/openvpn-client.key
User.txt	Username and password file	/config/cert/openvpn/openvpn-user.txt

You can configure the OpenVPN feature via the Web UI path: Network \rightarrow OpenVPN for the ALE Myriad Series phones.



If the username and password are downloaded as an auto-p file, should use the following format:

- unix file format
- username on the first line and the password on the second line, for example:

```
$
    cat /config/cert/openvpn/openvpn-user.txt
username
password$
```

• Inside the OPENVPN configuration files need to add a line: auth-user-pass /config/cert/openvpn/openvpn-user.txt

The following table lists the parameters you can use to configure openvpn.

Parameter	DeviceNetworkOpenVpnEnable	config.xml
Description	It configures openvpn switch.	
Permitted Values	false:disable true:enable	
Default	false	
Web UI	NetWork → OpenVPN → Enable	
Parameter	DeviceNetworkOpenVpnServerAddr	config.xml
Description	It configures OpenVPN server address.	
Permitted Values	String within 256 characters.	
Default	0.0.0.0	



Web UI	NetWork → OpenVPN → Server Address		
Parameter	DeviceNetworkOpenVpnServerPort	config.xml	
Description	It configures openvpn server port.		
Permitted Values	1-65535		
Default	1194		
Web UI	NetWork → OpenVPN → Server Port		
Parameter	DeviceNetworkOpenVpnTransport	config.xml	
Description	It configures openvpn transport protocot		
Permitted Values	UDP TCP		
Default	UDP		
Web UI	NetWork → OpenVPN → Transport Protocol.		
Parameter	DeviceNetworkOpenvpnAuthFileUrl	config.xml	
Description	It configures openvpn username and password file.		
Permitted Values	String within 256 characters. Note:Based on the openvpn standard format, the text is in unix formathe username, and the second line is the password.	at. The first line is	
Default	Blank		
Web UI	NetWork → OpenVPN → Username & Password		
Parameter	DeviceNetworkOpenvpnCaCertUrl	config.xml	
Description	It configures download url of openvpn ca file.		
Permitted Values	String within 256 characters.		
Default	BLANK		
Web UI	NetWork → OpenVPN → CA Certificate		
Parameter	DeviceNetworkOpenvpnClientCertUrl	config.xml	
Description	It configures download url of openvpn client cert file.		
Permitted Values	String within 256 characters.		
Default	BLANK		



Web UI	NetWork → OpenVPN → VPN Certificate	
Parameter	DeviceNetworkOpenvpnClientKeyUrl	config.xml
Description	It configures download url of OpenVPN client cert key file.	
Permitted Values	String within 256 characters.	
Default	BLANK	
Web UI	NetWork → OpenVPN → VPN Key Certificate	
Parameter	DeviceNetworkOpenvpnConfigFileUrl	config.xml
Description	It configures download url of openvpn config file.	
Permitted Values	String within 256 characters.	
Default	BLANK	
Web UI	NetWork → OpenVPN → VPN Configuration File	

3.5.2 Optimized headset button logic and enhanced experience

Parameter	FeatureHeadsetPriorEnable	config.xml
Description	It configures to enable the headset priority function.	
Permitted Values	false: Disable. true: Enable. Note: Channel high priority on headset, even when a channel switch of	occurs during a call.
Default	false	
Web UI	Features → General → Headset Prior:	

3.5.3 Optimized call waiting function to enable quick switching through programkey in the call interface

In R140, we have optimized the reminder of multiple calls. When there is an incoming call or talking, we will use 182 to respond to the second call when call waiting is enabled.

Parameter	FeatureKeepCallWaitingEnable	config.xml
Description	It configures to enable or disable Call Waiting after a call ends. After the function is activated, you can set programkey to call war flexible switch operation. You can also operate during a call, which flexible DND function.	•



Permitted Values	false: Disable the keep call waiting feature. true: Enable the keep call waiting feature.
Default	true

3.5.4 Optimized the ring tone for automatic answering

Parameter	FeatureAutoAnswerDelay	config.xml
Description	The automatic call answer delay is set. Note: This parameter is only valid if AccountXAutoAnswerEnable	e = true.
Permitted Values	1-60, Unit: second	
Default	1	
Web UI	None	
Phone UI	None	
Parameter	FeatureAutoAnswerToneEnable	config.xml
Description	It configures the auto answer whether the prompt tone is played	d.
Permitted Values	true - enable false - disable	
Default	false	
Web UI	None	_
Phone UI	None	

3.5.5 Optimized the logic of missed call and voice mail pop-up prompts





3.5.6 Optimized display of extended information during calls based on ldap

Scenario:

Employee: ALETEST1; number 10012; Position: Test Engineer.

Default behavior and optimization behavior (LdapExtraDisplay set as title)





Parameter	LdapExtraDisplay	config.xml		
Configure additional LDAP attributes to be displayed on the conversation.				
Dogginting	After the configuration, additional properties are displayed after the screen and call screen.	name of the call		
Description	Support key: sn; telephoneNumber; givenName; mobileNumber; otherNumber name; officephone; department; title.			
	Note: If you want to configure more parameters, you need user ",".			
Permitted Values	String within 126 characters			
Default	Blank			
Web UI	NA			

3.5.7 Optimized the default account display logic

When multiple accounts are configured, you can specify the default account to set the default outgoing number and modify the display policy on home bar to display the default account information.

Note The phone support number 1~8 for M3/M5/M7, and 1-20 for M8.

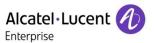
Parameter	SIPDefaultAccount	config.xml
Description	It configures the SIP phone default account.	
Permitted Values	1 - Account 1 2 - Account 2	

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	3 - Account 3		
	4 - Account 4		
	5 - Account 5		
	6 - Account 6		
	7 - Account 7		
	8 - Account 8		
	9 - Account 9		
	10 - Account 10		
	11 - Account 11		
	12 - Account 12		
	13 - Account 13		
	14 - Account 14		
	15 - Account 15		
	16 - Account 16		
	17 - Account 17		
	18 - Account 18		
	19 - Account 19		
	20 - Account 20		
Default	1		
Web UI	features → sip → Default Account		
Parameter	SettingDefaultAccountDisplayMode	config.xml	
	It configures set whether to display default account information on		
Description	home bar.		
Permitted	0 - Do not display default account on home bar.		
Values	1 - Display default account on home bar.		
Default	0		
Web UI	None		



The following is an example of whether the default account is displayed on home bar.



3.5.8 Optimized Increase the channel of Multicast Paging by 0, changing the channel range to 0-25

3.5.9 Optimized action uri support AOM

Variable Value	Phone Action
(F_) 0-9/*/ F_STAR/F_POUND	Short press the number key
(F_) E{x}_{y}	X=AOM index y=Corresponding AOM key. For example: E1_1 = AOM 1 first key E2_2 = AOM 2 second key
F_ E{x}_{y}_LONGPRESS	X=AOM index y=Corresponding AOM key. For example: E1_1_LONGPRESS = Long press AOM 1 first key E2_2_LONGPRESS = Long press AOM 2 second key
E{x}_LEFT、E{x}_RIGHT、E{x}_HOME	X=AOM index. Press the bottom three buttons above the AOM.

- 3.5.10 Optimize the discovery mode of Bluetooth, only in the Bluetooth interface can be discovered
- 3.5.11 Optimized USB flash drive supports NTFS format and FAT32 format



4 Bug Fixes

List of bugs fixed
No ethernet Link pop-up after synchronizing config file
The css panic occurs when RTCP is received during a multi-channel call
Need to support a shorter Register Retry Time
Audio delay in re-invite come in
It's slow to log in website
Failover/Fallback not working as expected
Need to support XSI call log display as number when username is Unavailable
MWI not working with Epygi platform
Semi-Attended transfer not working fine
Cannot return back to Login page after resetting factory on WEB
[XML Browser]The last line of the screen interface is incomplete
M8 won't send DNS query if failed once
The status of M7's BLF is not correct after connecting EM200
OPENVPN vpn file downloads should not depend on config changes
Support importing OpenVPN certificates and configuration files at one time
DND status is incorrect on the phone screen



5 New Features Descriptions

5.1 Added support local conference manage

Support for local conference management capabilities, the ability to control hold and mute for each participant.



5.2 Added support LDAP multi-line display support

LDAP custom call display policy

Scenario:

Employee: ALETEST1; number 10012; Position: Test Engineer.

Default behavior and optimization behavior (LdapExtraDisplay set as title)





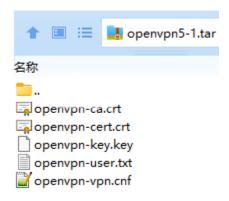
Parameter	LdapFieldsMapping	config.xml
Description	It configures additional LDAP attributes to be displayed on the power module page. It configures multiple attributes separated by commas (,), such as department and title. Its configuration, additional properties are displayed after the name of the call screen and the name of the call screen. The display sequence is based on the configured value.	
Permitted Values	String within 126 characters.	
Default	BLANK	



5.3 Added support OpenVPN tar file import

Compressed packet mode:

You can refer to the following example, after all the files of the vpn are named according to the convention, the unified compression into a tar file, and then upload the compressed package through the autop way, you can import it at one time.

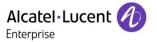


Parameter	DeviceNetworkOpenvpnUrl	config.xml
	Add a compressed package to import all OpenVPN certificates and files at one time to complete deployment.	
	The compressed package contains CA certificate, VPN certificate, VPI configuration file, and auth file.	N private key, VPN
The file name must be strictly matched:		
Description	openvpn-ca.crt	
	openvpn-cert.crt	
	openvpn-key.key	
	openvpn-user.txt	
	openvpn-vpn.cnf	
Permitted Values	String within 256 characters.	
Default	BLANK	

5.4 Added support separate wireless and wired parameters

Split wired and wireless configuration parameters to suit more scenarios.

Wired	Wifi
DeviceNetworkDns1	DeviceWifiDns1
DeviceNetworkDns2	DeviceWifiDns2
DeviceNetworkDns3	DeviceWifiDns3



DeviceNetworkIpv6Dns1	DeviceWifilpv6Dns1
DeviceNetworkIpv6Dns2	DeviceWifilpv6Dns2
DeviceNetworkIpv6Dns3	DeviceWifilpv6Dns3
DeviceNetworklpStackMode	DeviceWifilpStackMode
DeviceNetworkDhcpMode	DeviceWifiDhcpMode
DeviceNetworklpAddress	DeviceWifilpAddress
DeviceNetworkIpv6Address	DeviceWifilpv6Address
DeviceNetworkSubnetMask	DeviceWifiSubnetMask
DeviceNetworkGateway	DeviceWifiGateway
DeviceNetworklpv6DhcpMode	DeviceWifilpv6DhcpMode
DeviceNetworklpv6PrefixLen	DeviceWifilpv6PrefixLen
DeviceNetworklpv6Gateway	DeviceWifilpv6Gateway
DeviceNetworkStaticDnsEnable	DeviceWifiStaticDnsEnable

5.5 Added G726 codec support for M3/M5/M7/M8/H3X6

Added M3/M5/M7/M8/H3X6 support for the G726 codec.

5.6 Added support time synchronization based on SIP signaling

In addition to NTP synchronization time, we also added support for SIP signaling time synchronization. The phone used the time in sip signaling as the time to be synchronized through the 200 OK carried by the corresponding date header field returned by the server during registration.

The following table lists the parameters you can use to configure the NTP.

Parameter	SettingTimeMethod	config.xml
Description	Configure time synchronization through the NTP server or SIP signaling or manually set the time.	
Permitted Values	0: SNTP (default) 1: SIP Server 2: Manual	
Default	0	
Web UI	Settings → Time & Date → TimeMethod	



5.7 Added support new language Thai for MMI, Russian and Turkish for WBM

5.8 Added support on-hook to transfer function

Parameter	FeatureBlindTransferOnHookEnable	config.xml
Description	Enable and disable the blind transfer operation when hanging up.	
Permitted	false: disable feature	
Values	true: enable feature	
Default	true	
Parameter	FeatureAttendedTransferOnHookEnable	config.xml
Description	Enable and disable the attended transfer operation when hanging up.	
Permitted	false:disable feature	
Values	true:enable feature	
Default	true	
Parameter	FeatureTransAfterConfEnable	config.xml
	It configures the meeting initiator to hang up the phone and allow the	
	other two parties to continue the conversation.	
Description	Note: It is takes effect only for Local conferences.	
Permitted	false:disable feature	
Values	true:enable feature	
Default	false	
Parameter	FeatureTransProgKeyDealType	config.xml
Description	It configures the behavior after the Programmable key is pressed during	g a call.
	0: New Call	
	1: Attended Transfer	
Permitted	2: Blind Transfer	
Values	3: Blind Transfer Optional	
Default	2	

Option Transfer call: When DUT configuration FeatureTransProgKeyDealType = 3, call press BLF key, pop-up function selection interface (does not affect the current call status), can use the direction key + the Select (OK) key choose corresponding call operation; Or click Cancel (C key) to exit the selection screen and return to the call screen (Hard keys for other functions do not take effect).





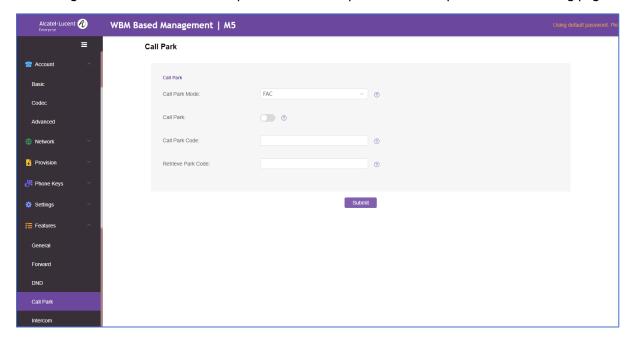
5.9 Added support set ringtones for contacts



5.10 Added support control to automatically generate a call park softkey on the call page

Call Park configuration and interface show

You configure information about the prefabricated call park or retrieve park on the following page:





• Configure FeatureCallParkEnable is true:



• Configure FeatureCallParkEnable is false:



Parameter	FeatureCallParkEnable	config.xml
Description	It configures the Call Park function is enabled or disabled. After activation, the call park button is displayed on the softkey on the talking.	
Permitted	false:disable	
Values	true:enable	
Default	false	
Web UI	Features → Call Park → Call Park	
Parameter	FeatureCallParkParkCode	config.xml
Description	It configures the call park code.	



Permitted Values	String within 64 characters.	
Default	Blank	
Web UI	Features → Call Park → Call Park Code	
Parameter	FeatureCallParkRetrieveCode config.xml	
Description	It configures the call park retrieve code.	
Permitted Values	String within 64 characters	
Default	Blank	
Web UI	Features → Call Park → Retrieve Park Code	
Parameter	FeatureCallParkDirectCallEnable	config.xml
Description	It configures enable or disable whether to make a direct call after pressing the Park/Retrieve softkey. This parameter is valid only when FeatureCallParkMode = 0 and the CallPark/Retrieve code is configured	
Permitted Values	false: disable true: enable	
Default	true	

5.11 Added support recording related functions, for file upload, automatic cleaning support

ALE phones support manual recording during a call or automatic recording once the call is set up.

Before recording, ensure that the USB disk has been connected to the IP phone. This is for devices that have USB A ports on them. Currently, support FAT32 and NFTS format. Supports a maximum of 64 GB memory space.

Parameter	FeatureUsbCallRecordingEnable	config.xml
Description	It enables or disables the call recording (using a USB flash drive) feature of the IP phone.	
Permitted Values	false - disable true - enable	
Default	false	
Parameter	FeatureAutoRecordingEnable	config.xml



Description	It enables or disables the automatic recording feature of the IP phone.	
Permitted	false - disable	
Values	true - enable	
Default	false	

ALE phones support backup recording file to server.

After the upload function is enabled, the phone can use http or https to upload files using put or post to back up files to the server. It also supports automatic upload and manual upload for easy operation.

Parameter	FeatureRecordingUploadEnable	config.xml
Description	It configures to enable the local automatic recording upload function.	
Permitted	false: Disable the phone recording file upload function.	
Values	true: Enable the phone recording file upload function	
Default	false	
Parameter	FeatureRecordingUploadServerUrl	config.xml
Description	It configures the IP address of the server for uploading local recording fi	les is specified
Permitted	URL within 511 characters	
Values	Note: It is valid only the FeatureRecordingUploadEnable is true.	
Default	Blank	
Parameter	FeatureRecordingUploadServerUsername	config.xml
	It configures the authentication username of the server where the local recording file is	
Description	uploaded is specified.	
	Note: It is valid only the FeatureRecordingUploadEnable is true.	
Permitted Values	String within 64 characters	
Default	Blank	
Parameter	FeatureRecordingUploadServerPassword	config.xml
	It configures the authentication password of the server where the local recording file is	
Description	uploaded is specified.	
	Note: It is valid only the FeatureRecordingUploadEnable is true.	
Permitted Values	String within 64 characters	
Default	Blank	

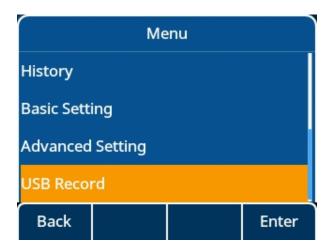


Parameter	FeatureRecordingUploadAutoEnable	config.xml
Description	It configures to enable the automatic uploading of local recording files to the server immediately after the generation of local recording files.	
Permitted Values	false: Disable the phone immediately auto recording file upload function. true: Enable the phone immediately auto recording file upload function. Note: It is valid only the FeatureRecordingUploadEnable is true.	
Default	false	
Parameter	FeatureRecordingUploadDailyEnable	config.xml
Description	It configures to enable the scheduled upload of local recording files.	
Permitted Values	false: Disable the phone recording file scheduled upload function. true: Enable the phone recording file scheduled upload function. Note: It is valid only the FeatureRecordingUploadEnable is true.	
Default	false	
Parameter	FeatureRecordingUploadBeginTime config.xml	
Description	It configures to enable the scheduled upload of local recording files start time.	
Permitted Values	Time from 00:00 to 23:59 Note: It is valid only the eatureRecordingUploadDailyEnable is true.	
Default	00:00	
Parameter	FeatureRecordingUploadEndTime	config.xml
Description	It configures to enable the scheduled upload of local recording files end	time.
Permitted Values	Time from 00:00 to 23:59 Note: It is valid only the FeatureRecordingUploadDailyEnable is true.	
Default	00:00	
Parameter	FeatureRecordingAutoDeleteEnable	config.xml
Description	It configures enable or disable the function of automatically deleting USB flash drive recording files.	
Permitted Values	false - Disable true - Enable	
Default	false	
Parameter	FeatureRecordingAutoDeleteThreshold	config.xml
Description	It configures the remaining capacity of the USB flash drive, the earliest	-



	recording files are automatically deleted.	
Permitted Values	The integer ranges from 0 to 1024. The unit is MB.	
Default	20	
Parameter	FeatureRecordingFileDeleteMethod	config.xml
Description	It configures the method for automatically deleting recording files. Delete or not delete recording files after uploading successfully.	
Permitted Values	0: Do not delete local recording files.1: Delete recording files after uploading successfully.Note: It is valid only the FeatureRecordingUploadEnable is true.	
Default	0	
Parameter	FeatureRecordingUploadRetryTimes	config.xml
Description	It configures retry times when recording file upload failed.	
Permitted Values	0-5 Note: zero is meads that do not retry. Note: It is valid only the FeatureRecordingUploadEnable is true.	
Default	2	

When you insert the USB drive, you can see a USB directory appears under the Menu item. This directory can access the recording files generated during your call.



On this screen, you can view the recording file in wav format. The recording file contains the call time and the called and called account information as the file name.

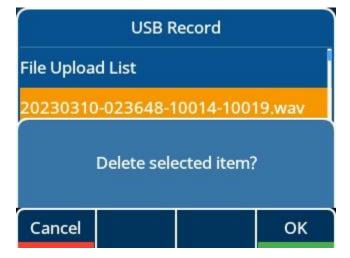


ALE Phones maximum of 1000 entries are supported.



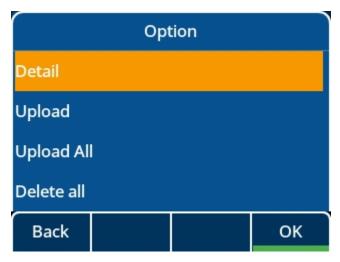
When you select the relevant recording information, we can see that softkey supports delete, details, and play functions.

• You press "Delete" key to delete a record.

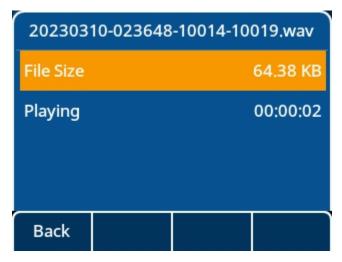




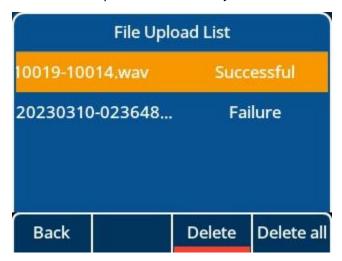
• You can press "Option" key to view more option information, for example: Delete all, Upload, Detail......



• You can press "Play" to play record file.



• The "File Upload List" will tell you success or fail the record file upload to server.





If the upload takes a while, you can see an upload icon in idle, indicating that the upload is in progress.



5.12 Added support CDP

Cisco Discovery Protocol (CDP) is a private binary-layer networking protocol developed by Cisco. It is automatically loaded by most Cisco devices upon startup. By using CDP, Cisco devices can share information such as operating system software version, device identifiers, address tables, port identifiers, and performance metrics among themselves and their direct connected devices.

Like HP's LLDP and Huawei/H3C's NDP protocols, CDP uses a set of rules and filters to discover and enumerate all network devices on the local network. The main difference between CDP and these other protocols is that CDP provides a more private and secure way to discover and enumerate network devices on a local network.

In addition to the direct sharing of device information, CDP also supports the discovery of other network devices. When a device is connected to a network, it notifies the local device of the presence of other network devices and allows the local device to discover these devices. This allows for a more seamless and secure network operation.

Overall, CDP is a powerful networking protocol that allows Cisco devices to work together more seamlessly and securely on the local network.

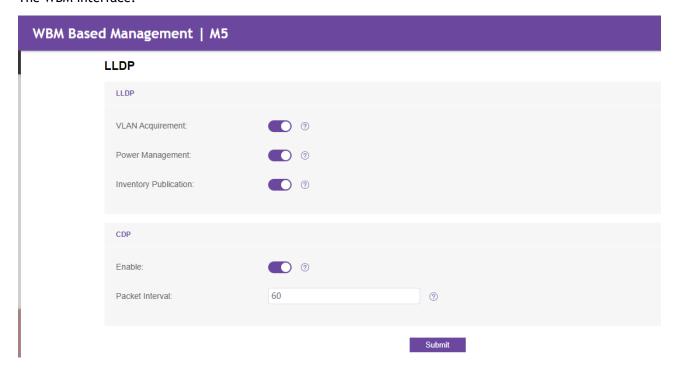
The following table lists the parameters you can use to configure CDP.

Parameter	DeviceNetworkCdpEnable	config.xml
Description	It enables or disables the CDP (Cisco Discovery Protocol) feature on the IP phone.	
Permitted Values	true - enable false - disable	
Default	true	



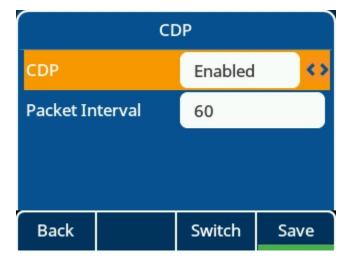
Web UI	Network → LLDP → CDP → Enable	
Phone UI	Menu → Advanced Setting (default password: 123456) → Network → CDP→ enable	
Parameter	DeviceNetworkCdpPacketInterval	config.xml
Description	It configures the interval for sending CDP packets	
Permitted Values	1-3600 seconds	
Default	60	
Web UI	Network → LLDP → CDP → Packet Interval	
Phone UI	Menu → Advanced Setting (default password: 123456) → Network → CDP → Packet Interval	

The WBM interface:

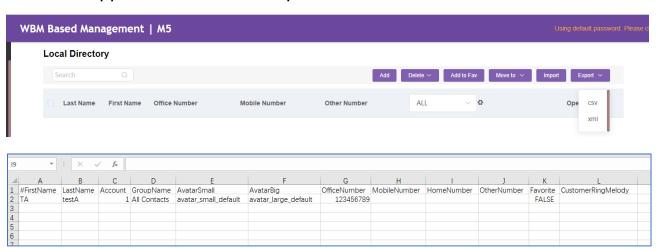




The MMI interface:



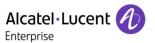
5.13 Added support the contact template format can be csv



5.14 Added support the different refuse code for different scenarios

When traffic is busy in different scenarios, ALE phones can customize refused code in different scenarios so that the peer end can receive customized information.

Parameter	FeatureNormalRefuseCode	config.xml
	It configures the return code for SIP response messages in the case of call rejection.	
	Scenario coverage: Manually reject incoming calls, automatically reject incoming	
	calls from blacklisted numbers, and automatically reject incoming calls caused by	
Description	max call or call waiting off.	



	404 404/N-4 5		
	404 - 404(Not Found)		
	480 - 480(Temporarily Unavailable) 486 - 486(Busy Here)		
	600 - 600(Busy Everywhere)		
Permitted Values	603 - 603(Decline)		
Default	486		
Web UI	None		
Parameter	FeatureDndRefuseCode	config.xml	
	It configures the return code of SIP response messages when DND enable		
Description	rejects incoming calls.		
	404 - 404(Not Found)		
	480 - 480(Temporarily Unavailable)		
	486 - 486(Busy Here)		
Permitted	600 - 600(Busy Everywhere)		
Values	603 - 603(Decline)		
Default	486		
Web UI	None		
Parameter	FeatureNoAnswerCode	config.xml	
Description	It configures the return code for SIP response messages when no answer	times out.	
	404 - 404(Not Found)		
	480 - 480(Temporarily Unavailable)		
	486 - 486(Busy Here)		
Permitted	600 - 600(Busy Everywhere)		
Values	603 - 603(Decline)		
Default	486		
Web UI	None		

5.15 Added support port setting capability for pnp and ssh

Parameter	DeviceProvisionPnpPort	config.xml
Description	Configure pnp srouce port.	
Permitted Values	Integer from 1 to 65535	



Default	5082	
Web UI	None	
Parameter	DeviceSecuritySshPort	config.xml
Description	It configures the ssh port.	
Permitted Values	0-65535	
Default	22	
Web UI	None	

5.16 Added support custom tone

Although we have prefabricated a complete set of audio configuration options for different regions, you may want to customize your own audio to be more local, such as different frequencies, different duration, custom customer audio is a good way to meet this requirement.

Custom Tone example:

Tone = Freq/Duration[; Freq/Duration][; Freq/Duration]...

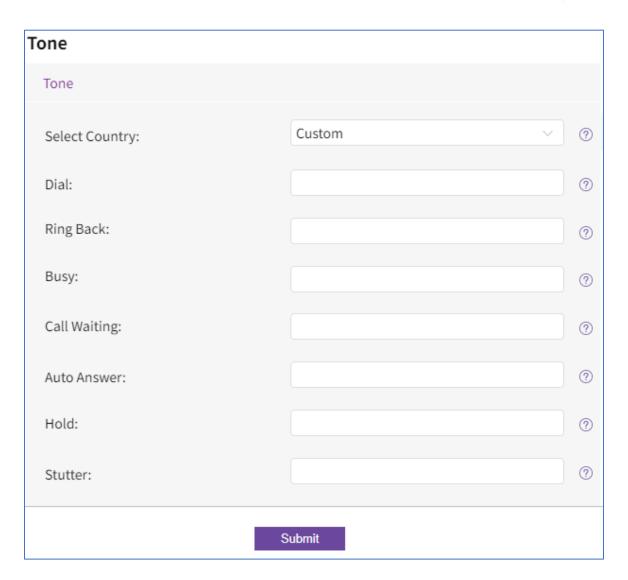
Freq = Freq1[+Freq2] [+Freq3][+Freq4]

Description:

- Freq/Duration indicates a group of frequencies. A Tone can contain a maximum of eight frequencies.
- A group of frequencies supports the juxtaposition of a maximum of two frequencies.
- Freq ranges from 200 to 4000 Hz, Duration ranges from 0 to 30000ms; Freq=0 indicates that the mute Duration is not played. You can set the mute duration by 0/Duration.
- Setting example: SettingDialTone = 200/1000; 0/1000; 200+2000/1000: After the Dial Tone is triggered, the frequency of 200Hz is played for 1s, and then the frequency of 200Hz and 2000Hz is muted for 1s, and then the two frequencies are played for 1s at the same time.
- Special value description:
 - Freq without /Duration indicates the frequency of the group. For example, 200/1000; 300 means that the frequency of 200Hz is played for 1s, and then the frequency of 300Hz is played all the way.
 - Tone with "!" It is played only once. For example: 200/1000; 0/1000; 200+300/1000, which
 means to play 1s at 200Hz, then mute 1s, then play 1s at both 200Hz and 300Hz, and then
 end Tone play.

WEB interface: Path: Setting-Audio



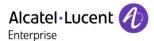


The following table lists the parameters you can use to configure tones.

Parameter	SettingDialTone	config.xml
Description	Configure dial tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	
Default	Blank	
Parameter	SettingSecondaryDialTone	config.xml
Description	Configure secondary dial tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	



Default	Blank	
Parameter	SettingRingBackTone	config.xml
Description	Configure ring back tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	
Default	Blank	
Parameter	SettingBusyTone	config.xml
Description	Configure ring back tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	
Default	Blank	
Parameter	SettingCongestionTone	config.xml
Description	Configure congestion tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	
Default	Blank	
Parameter	SettingCallWaitingTone	config.xml
Description	Configure call waiting tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	
Default	Blank	
Parameter	SettingRecallDialTone	config.xml
Description	Configure recall dial tone.	
Permitted Values	String within 511 characters. Invalid value does not take effect.	
Default	Blank	
Parameter	SettingStutterTone	config.xml
Description	Configure stutter tone.	1
Permitted	String within 511 characters.	
Values	Invalid value does not take effect.	
Default	Blank	



Parameter	SettingAutoAnswerTone	config.xml
Description	Configure auto answer tone.	
Permitted	String within 511 characters.	
Values	Invalid value does not take effect.	
Default	Blank	
Parameter	SettingMessageTone	config.xml
Description	Configure Message tone.	
Permitted	String within 511 characters.	
Values	Invalid value does not take effect.	
Default	Blank	
Parameter	SettingSpecialInfoTone	config.xml
Description	Configure special information tone.	
Permitted	String within 511 characters.	
Values	Invalid value does not take effect.	
Default	Blank	

5.17 Added support codec G726 for M8

5.18 Added support call display source

This section describes how to configure the call display source for ALE myriad phones. Users can select user-defined rules to determine whether to use local contact information or sip signaling contact information when making a call. The priorities for local contact information are as follows: Local Directory>Remote Phone Book>LDAP Directory>History

Note

X means account ID. It can be number 1-8 for M3/M5/M7, and 1-20 for M8.

Parameter	CallDisplaySource	config.xml
Description	It configures a name matching priority policy for phone calls.	
Permitted Values	0 - Local Directory>Remote Phone Book>Network Contacts>LDAP Directory>Network signaling 1 - Network signaling / SIP signaling	ory>Enterprise
Default	0	



Parameter FeatureDiversionInfoEnable It configures whether to display an incoming call with the Diversion header information. This scenario is mostly when the number is forwarded. Permitted Values false: Do not display via information. True: Display via information. Default true Web UI None Parameter AccountXCallerSource configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed Permitted Values 2: From/To Default 0;1;2 Web UI None Parameter AccountXCalleeSource configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed 0: PAI 1: RPID 2: From/To Default 0;1;2 Web UI None Parameter AccountXCalleeSource configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed O: PAI 1: RPID 2: From/To Description repeating is allowed 0: PAI 1: RPID 2: From/To Description repeating is allowed 0: PAI 1: RPID 2: From/To Default 0;1;2 Web UI None	Web UI	None		
Description This scenario is mostly when the number is forwarded. Permitted Values true: Display via information. Default true Web UI None Parameter AccountXCallerSource config.xml When a SIP phone receives an incoming call, it obtains the peer information from the corresponding fields in the configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed 0: PAI 1: RPID Values 2: From/To Default O;1;2 Web UI None Parameter AccountXCalleeSource config.xml When a SIP phone sends an outing call, it obtains the peer information from the corresponding fields in the configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed 0: PAI 1: RPID 2: From/To Description repeating is allowed 0: PAI 1: RPID 2: From/To Default 0;1;2	Parameter	FeatureDiversionInfoEnable config.xml		
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Web UI Parameter AccountXCallerSource When a SIP phone receives an incoming call, it obtains the peer information from the corresponding fields in the configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed 0: PAI Permitted 1: RPID 2: From/To Default 0;1;2 Web UI None Parameter AccountXCalleeSource config.xml When a SIP phone sends an outing call, it obtains the peer information from the corresponding fields in the configured header priority sequence and displays the information. Note: three parameters need to be sorted in order of priority, and no missing or repeating is allowed 0: PAI Permitted 1: RPID 2: From/To Default 0;1;2	Values	true: Display via information.		
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Description repeating is allowed 0: PAI Permitted 1: RPID Values 2: From/To Default 0;1;2		displays the information.		
O: PAI 1: RPID 2: From/To Default 0;1;2		Note: three parameters need to be sorted in order of priority, and no missing or		
Permitted 1: RPID 2: From/To Default 0;1;2	Description	repeating is allowed		
Values 2: From/To Default 0;1;2		0: PAI		
Default 0;1;2	Permitted	1: RPID		
	Values	2: From/To		
Web UI None	Default	0;1;2		
	Web UI	None		



5.19 Added support customer language

To better satisfy the user's local language configuration and add user-defined functions, the current version has supported the customer language feature, user can modify or add the language.

Format:

- It should be noted that the file is a .txt format file.
- The inner separator is the tab key, not the space key. example: tab is between S_PLACED_CALLS and Chinese_simplified0.
- Only be one translation in a row.

Translation_Source CountryName Translation_Value

Example:

S_PLACED_CALLS	French	AppelsAAA passés
S_PLACED_CALLS	Chinese	_simplified 呼叫
S_PLACED_CALLS		
S_PLACED_CALLS	Custom	Call

Description:

- S_PLACED_CALLS: This is the translation source, which corresponds to the id of the phone in it.
- French/Chinese simplified: This is the language option identifier, which identifies the type of language that needs to be changed. In the example, four languages were changed.
- AppelsAAA passes / Place call: This is the last character you want to translate. The spacing between characters in this option is space, not tab.

The following table lists the parameters you can use to configure the phone.

Parameter	SettingCustomLanguageUploadUrl	config.xml
Description	It configures phone download custom language file url.	
Permitted Values	Configure a valid URL address format. Note: The function base parameter is set to Setting Language=99	
Default	BLANK	
Phone UI	None	



Submitting a Service Request

Please connect to our eService Request application.

Before submitting a Service Request, please be sure:

The application has been certified via the AAPP if a third party application is involved.

You have read the release notes that list new features, system requirements, restrictions, and more, and are available in the <u>Technical Documentation Library</u>.

You have read through the related troubleshooting guides and technical bulletins available in the <u>Technical</u> <u>Documentation Library</u>.

You have read through the self-service information on commonly asked support questions and known issues and workarounds available in the <u>Technical Knowledge Center</u>.

- END OF DOCUMENT -