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Troubleshooting Guide FOR Alcatel-Lucent Enterprise SIP DeskPhones

This document provides the FAQs for Alcatel-Lucent Enterprise Myriad and Halo series DeskPhones.

Revision History

Edition 1: September 12, 2024	creation of the document
Edition 2: October 28, 2024	update Q&A 27/28, add Q&A 33
Edition 3: January 9, 2025	remove H2/H3P

Table of contents

1 Phone can't be powered up or reboot cycled

1.1 Issue Summary

Customers may meet the problem that when connected the phone with power supply, the phone can’t be powered up or reboot cycled.

The Deployment manager may also need to configure the POE switch to make sure every phone can boot up fine.

1.2 Possible Causes

The power supply is not corrected which may cause this issue.

1.3 How to Resolve

ALE SIP DeskPhones support IEEE 802.3af standard and compatible with devices that applied with IEEE 802.3af standard as well. Please see below data:

ALE SIP DeskPhone Power Conclusion					
	External Power adaptor		POE		Class
	Input	Max(W)	Idle(W)	Max(W)	(IEEE802.3af)
H2P	5V/0.6A	1.65W	2.11W	3.22W	Class1
H3G	5V/2A	2.425W	0.98W	3.2W	Class1
H6	5V/2A	6.51W	1.1W	4.6W	Class2
M3	5V/2A	2.58W	2.17W	4.64W	Class2
M5	5V/2A	2.58W	2.17W	4.64W	Class2
M7	5V/2A	2.58W	2.17W	4.64W	Class2
M8	5V/2A	7.46W	2.31W	9.53W	Class3

1.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

1.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

1.6 Firmware Version

All version

2 How to configure IP address for an ALE SIP DeskPhone

2.1 Issue Summary

All IP phones need to get an IP address before using it, this FAQ shares how to configure IP address for an ALE SIP DeskPhone.

2.2 Possible Causes

Phone set up, basic IP configuration

2.3 How to Resolve

ALE SIP DeskPhones support three methods to set up an IP address:

Dynamic/Static/Alcatel dyn for H3G/H6/Myriad series DeskPhones

DHCP/Static IP/PPPoE for H2P DeskPhone

For DHCP/Dynamic mode, the phone will send out DHCP discover message to the DHCP server to get an IP address automatically, just make sure the LLDP setting is configured correctly, for phone side, no special settings needed, and all phones use this model by default.

Method to enable/disable the LLDP feature:

Phone UI	H2P: Setting -> Admin -> IP Param -> IP Config -> LLDP H3G/H6/Myriad series: Menu -> Advanced Setting -> Network -> LLDP -> VLAN Acquirement
Web UI	H2P: Network -> Advanced -> Link Layer Discovery Protocol (LLDP) Settings -> Enable LLDP H3G/H6/Myriad series: Network -> LLDP&CDP -> VLAN Acquirement

For Static mode, you need to get the IP address from your IT manager and configure manually to the phone side, normally, it is configured on phone side

H2P:

1. Press “Setting” -> “Admin” (password 123456) -> “IP Param” -> “IP Config” -> “IPv4 Settings”
2. Switch the Connection Mode to “Static IP”, then fill in the corresponding IP information accordingly, then press “OK” to save the configuration

For PPPoE mode just for H2P DeskPhones, press “Setting” -> “Admin” (password 123456) -> “IP Param” -> “IP Config” -> “IPv4 Settings”. Switch the Connection Mode to “PPPoE”, then fill in the corresponding user name and password accordingly, then press “OK” to save the configuration

H3G/H6/Myriad Series:

1. Press “Menu” -> “Advance Setting” (password 123456) -> “Network” -> “LAN Port” -> “IP Config” -> “IPv4 Settings”
2. Switch the IPv4 Mode to “Static”, then fill in the corresponding IP information accordingly, then press “OK” to save the configuration

Note All IP configuration change will cause the phone to reboot automatically.

2.4 More

For H2P DeskPhones IP Configuration, you can also refer to the H2/H2P DeskPhone - SIP Phones Deployment Guide with Cloud PBXs from Third Party Vendors (**Chapter 3.2 Configuring IP parameters and SIP account parameters via MMI** and **Chapter 3.3 Configuring IP parameters and SIP account parameters via WBM**) for more information.

For H3G/H6 and Myriad series DeskPhones IP Configuration, you can also refer to the Administration Manual for Myriad and Halo Series DeskPhone (**Chapter 2.1 IPv4 and IPv6 Network Settings** and **Chapter 2.4.1 LLDP Configuration**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

2.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

2.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

3 How to configure VLAN settings for ALE SIP DeskPhones

3.1 Issue Summary

The purpose of VLAN configuration on the IP phone is to insert a tag with VLAN information to the packets generated by the IP phone. If VLAN configuration is needed but not configured correctly, the phone will not be able to get the IP address correctly.

This FAQ shares the different ways to configure the VLAN feature for ALE SIP DeskPhones.

3.2 Possible Causes

Users need to distinguish different VLANs for different devices

3.3 How to Resolve

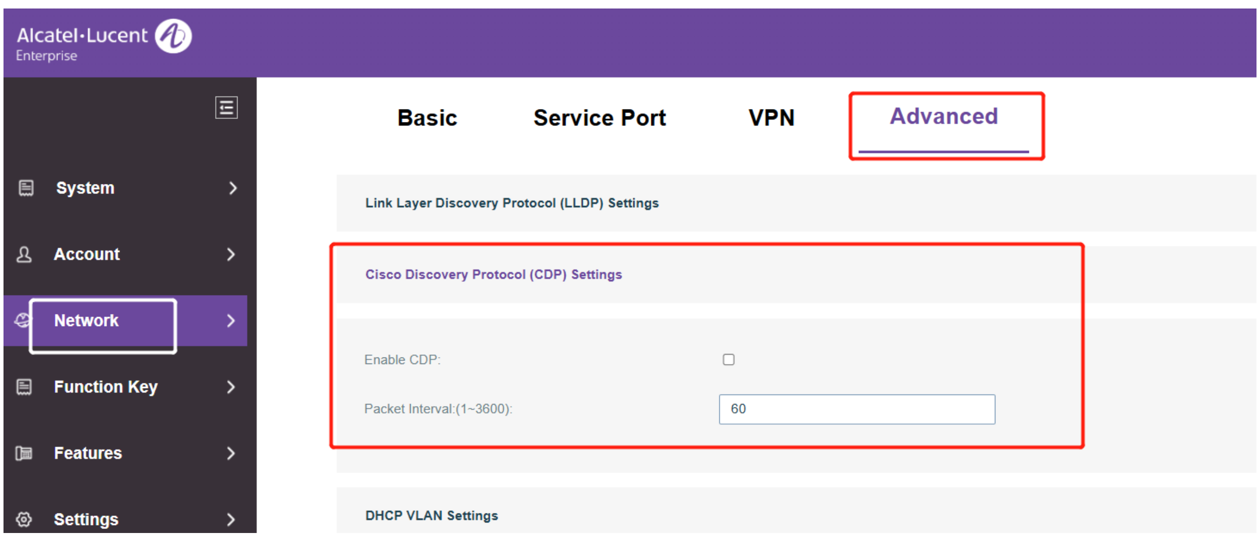
Customers can configure the VLAN via CDP, LLDP and DHCP automatically or set it manually.

CDP

H2P:

Phone UI: Not Supported

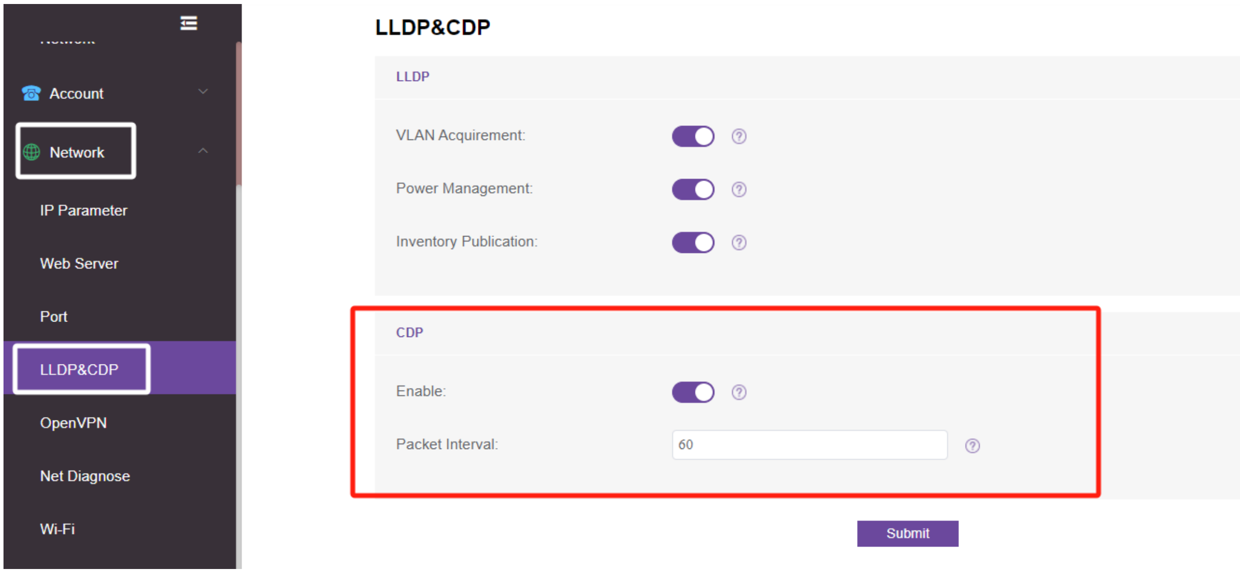
Web UI: Network -> Advanced -> CDP Settings



H3G/H6/Myriad Series:

Phone UI: Not Supported

Web UI: Network -> LLDP&CDP -> CDP



Auto provision & Phone UI: (H2P)

Parameter	CDPEnable
Description	It configures whether CDP is enabled.
Permitted Values	0- Disable. 1- Enable.
Default	0
Web UI	Network -> Advanced -> Cisco Discovery Protocol (CDP) Settings -> Enable CDP
Phone UI	Not Available
Parameter	**CDPRefreshTime**
Description	It configures the CDP requests interval time of the phone.
Permitted Values	1~3600
Default	60
Web UI	Network -> Advanced -> Cisco Discovery Protocol (CDP) Settings -> Packet Interval
Phone UI	Not Available

Auto provision & Phone UI: (H3G/H6/Myriad series)

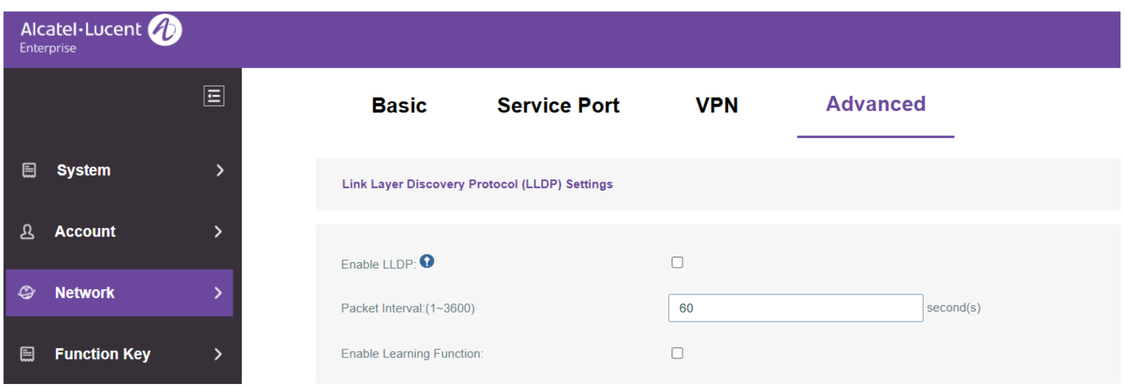
Parameter	DeviceNetworkCdpEnable
Description	It enables or disables the CDP (Cisco Discovery Protocol) feature on the IP phone.
Permitted Values	true - enable false - disable
Default	true
Web UI	Network -> LLDP&CDP -> CDP -> Enable
Phone UI	Menu -> Advanced Setting (default password: 123456) -> Network -> CDP -> CDP
Parameter	**DeviceNetworkCdpPacketInterval**
Description	It configures the interval for sending CDP packets.
Permitted Values	1-3600 seconds
Default	60
Web UI	Network -> LLDP&CDP -> CDP -> Packet Interval
Phone UI	Menu -> Advanced Setting (default password: 123456) -> Network -> CDP -> Packet Interval

LLDP

H2P:

Phone UI: Setting -> Admin -> IP Param -> IP Config -> LLDP

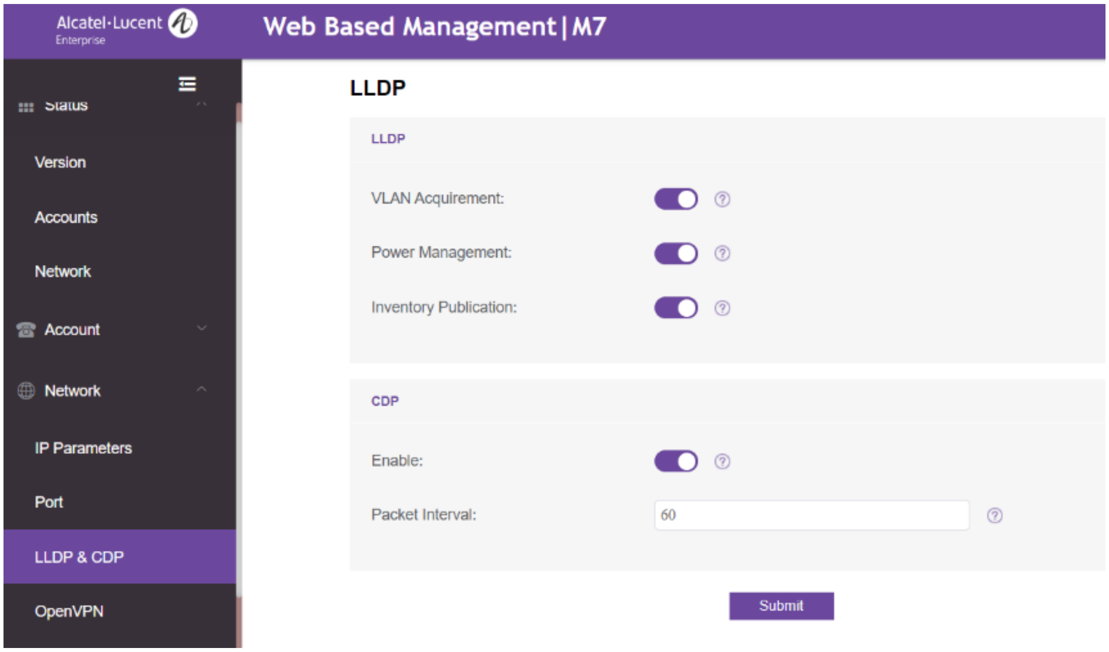
Web UI: Network -> Advanced -> LLDP



H3G/H6/Myriad series:

Phone UI: Menu -> Advanced Setting -> Network -> LLDP

Web UI: Network -> LLDP&CDP -> LLDP



Auto provision & Phone UI: (H2P)

Parameter	LLDPTransmit
Description	It configures whether LLDP is enabled.
Permitted Values	0- Disable. 1- Enable.
Default	1
Web UI	Network -> Advanced -> Link Layer Discovery Protocol (LLDP) Settings -> Enable LLDP
Phone UI	Setting -> Admin -> IP Param -> IP Config -> LLDP
Parameter	**LLDPRefreshTime**
Description	It configures the LLDP requests interval time of the phone.
Permitted Values	1~3600
Default	60
Web UI	Network -> Advanced -> Link Layer Discovery Protocol (LLDP) Settings -> Packet Interval
Phone UI	Not Available

Parameter	LLDPLearnPolicy
Description	It configures whether apply the learned VLAN ID to the phone configuration
Permitted Values	0- Disable. 1- Enable.
Default	1
Web UI	Network -> Advanced -> Link Layer Discovery Protocol (LLDP) Settings -> Enable Learning Function
Phone UI	Setting -> Admin -> IP Param -> IP Config -> LLDP -> Learning

Auto provision & Phone UI (H3G/H6/Myriad series)

Parameter	DeviceNetworkLldpVlanEnable
Description	It enables or disables the LLDP (Linker Layer Discovery Protocol) feature on the IP phone.
Permitted Values	true false
Default	true
Web UI	Network -> LLDP&CDP -> VLAN Acquirement
Phone UI	H3G/H6/Myriad series: Advanced Setting -> Network -> LLDP -> VLAN Acquirement

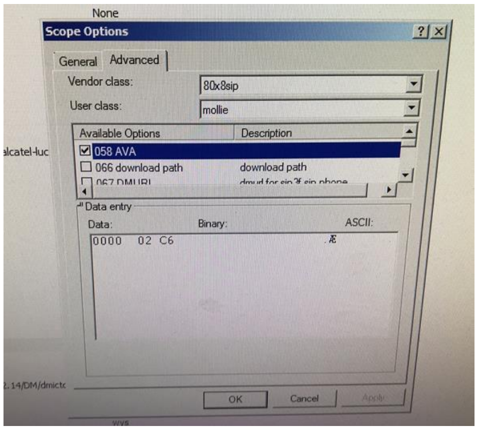
DHCP

H2P DeskPhones support pre-defined option 132 to carry the VLAN ID. You can also define the DHCP option used to carry VLAN ID, details below:

Parameter	DHCPOptionVlan
Description	It configures the DHCP Option get VLAN.
Permitted Values	128-254
Default	132
Web UI	Network -> Advanced -> DHCP VLAN Settings -> DHCP Option Vlan
Phone UI	Not Available

H3G/H6/Myriad series DeskPhones support VLAN discovery via DHCP. The predefined option 43 -> option 58 is used to supply the VLAN ID by default.

Here is an example of DHCP option 58 configuration for VLAN:

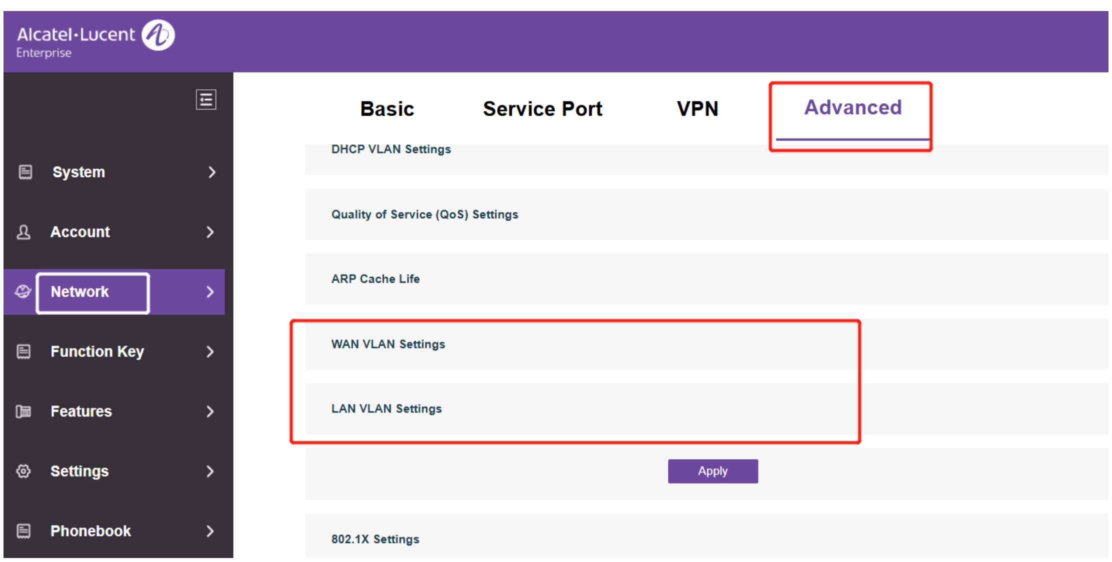


Configure VLAN Manually

H2P:

Phone UI: Setting -> Admin -> IP Param -> IP Config -> VLAN Config/LAN VLAN

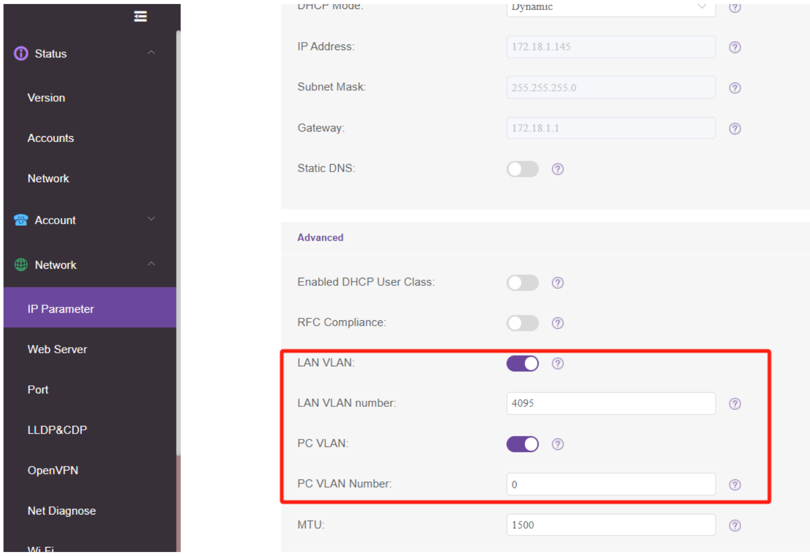
Web UI: Network -> Advanced -> WAN VLAN Settings/VAN VLAN Settings



H3G/H6/Myriad series:

Phone UI: Menu -> Advanced Setting -> Network -> Vlan -> VLAN Config/Data Vlan Config

Web UI : Nework -> IP Parameter -> LAN VLAN number/PC VLAN number



Auto provision & Phone UI: (H2P)

Parameter	EnableVLAN
Description	Enable VLAN to let Status access to VLAN network with vlan tagged
Permitted Values	0-Disable. 1-Enable.
Default	Disabled
Web UI	Network -> Advanced -> WAN VLAN Settings -> Enabled VLAN
Phone UI	Setting -> Admin -> IP param -> Vlan Config -> WAN VLAN
Parameter	**VLANID**
Description	LAN ID for Status WAN port
Permitted Values	Valid Value: 0~4095
Default	256
Web UI	Network -> Advanced -> WAN VLAN Settings -> WAN VLAN ID
Phone UI	Setting -> Admin -> IP param -> Vlan Config -> WAN VLAN ID
Parameter	**EnablePVID**
Description	It configures LAN port mode

Permitted Values	0- follow WAN 1- Disabled 2- Enabled
Default	follow WAN
Web UI	Network -> Advanced -> LAN VLAN Settings -> LAN VLAN Mode
Phone UI	Setting -> Admin -> IP param -> Vlan Config -> LAN VLAN
Parameter	**PVIDValue**
Description	It configures VLAN for the Internet (LAN) port.
Permitted Values	Valid Value: 0~4095
Default	254
Web UI	Network -> Advanced -> LAN VLAN Settings -> LAN VLAN ID
Phone UI	Setting -> Admin -> IP param -> Vlan Config -> LAN VLAN ID

Auto provision & Phone UI: (H3G/H6/Myriad series)

Parameter	**DeviceNetworkLanVlanEnable**
Description	It enables or disables the VLAN for the Internet port.
Permitted Values	true - enable false - disable
Default	false
Web UI	Network -> IP Parameters -> LAN VLAN
Phone UI	H3G/H6/Myriad series: Advanced Setting -> Network -> Vlan -> VLAN Config -> Use VLAN
Parameter	**DeviceNetworkLanVlanNumber**
Description	It configures the VLAN ID for the Internet port. Note: It works only if “LocalEnetcfgVlanEnable” is set to true.
Permitted Values	Integer from 1 to 4095
Default	4095
Web UI	Network -> IP Parameters -> LAN VLAN Number
Phone UI	H3G/H6/Myriad series: Advanced Setting -> Network -> Vlan -> VLAN Config -> ID

Parameter	DeviceNetworkPcVlanEnable
Description	It enables or disables the VLAN for the PC port.
Permitted Values	true - enable false - disable
Default	false
Web UI	Network -> IP Parameters -> PC VLAN
Phone UI	H3G/H6/Myriad series: Advanced Setting -> Network -> Vlan -> Data Vlan Config -> Use VLAN
Parameter	**DeviceNetworkPcVlanNumber**
Description	It configures the VLAN ID for the PC port. Note: It works only if "LocalEnetcfgDataVlanEnable" is set to true.
Permitted Values	Integer from 1 to 4095
Default	4095
Web UI	Network -> IP Parameters -> PC VLAN Number
Phone UI	H3G/H6/Myriad series: Advanced Setting -> Network -> Vlan -> Data Vlan Config -> ID

3.4 More

You can also refer to the [Administration Manual for ALE Myriad and Halo Series DeskPhone]() (**Chapter 2.4 VLAN**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

3.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

3.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

4 Phone can't get an IP address

4.1 Issue Summary

Customers may meet the problem that when connecting to a switch port, the phone can’t get an IP address automatically.

4.2 Possible Causes

Phone Hardware issue

Phone configuration issue

4.3 How to Resolve

1. Connection Test

Connect a network cable to the phone, if it doesn’t work, try the network cable in another network device. If the issue is the same, please check the cable and your network environment.

2. Hardware Test

Configure a static IP address to the phone and connect a PC to the phone PC port, then configure static IP address in PC with the same subnet, if the phone still can’t be accessed it should be a hardware problem. Please contact your vendor or local distributor and send the problem description for help.

For how to configure static IP, please refer to FAQ “How to configure IP address for an ALE SIP DeskPhone”

3. Configuration Issue

Try to do a factory reset to the phone and check again. There may be a wrong configuration in the phone. For the method to reset the phone, please refer to FAQ “How to reset the administrator password”

If still not work, please check the LLDP and VLAN options (Enable/Disable/VLAN ID) to see whether the issue can be solved or not, for more details, please refer to FAQ “How to configure VLAN settings for ALE SIP Deskphones”

4.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at

support.alesip@al-enterprise.com.

4.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

4.6 Firmware Version

All version

5 SIP account register failed

5.1 Issue Summary

Phone booted up, but on the LCD side, always show "No Service"

5.2 Possible Causes

Phone SIP account not configured or registered failed.

Common error:

1. User name or register name not correct
2. Password not correct
3. SIP server address or outbound server address not configured or incorrect
4. All account parameters are correct, but phone can't connect to PBX server side

5.3 How to Resolve

1. Check with your service provider and make sure all necessary SIP account parameters are correct, details below:

   User Name: It is account name provided by SIP PBX for registration

   Register Name: It is an authenticated ID provided by SIP PBX for registration

   Password: It is the authenticated key provided by SIP PBX for registration

   Server Host: It is the server address of PBX provided by SIP PBX for registration

   SIP Server Port: It is the register port provided by SIP PBX for registration

   Outbound Proxy Address: It is the outbound server address if needed

2. Please try to choose other transport mode (UDP/TCP/TLS) to see if issue solved or not
3. Please try to register another SIP account to see if it is the account issue.
4. Please check if your phone network can connect PBX normally or not. You can try to ping the server host to see if it is OK, if no, please connect your IT manager for help first

5.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

5.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

5.6 Firmware Version

All version

6 Phone registered and unregistered frequently

6.1 Issue Summary

Customers may meet the situation that the SIP account of the phone registered and after some time, it suddenly unregistered, then registered automatically, switch between them frequently which may cause the call failed at some time.

6.2 Possible Causes

1. Network not stable
2. Server side disabled the account or Phone SIP account configured incorrectly

6.3 How to Resolve

1. Check with your IT manager to solve the network issue

2. Set "Keep Alive" parameter to a shorter time like 30

   Web UI: (H2P)

      Account -> SIP -> Advanced Settings -> Keep Alive Interval

   Web UI: (H3G/H6/Myriad series)

      Account -> Advanced -> Keep Alive Timer

   Auto Provision & Phone UI: (H2P)

Parameter	sip.lineX.UDPUpdateTTL
Description	It configures the keep alive timer, X = 1-2
Permitted Values	[0,*]
Default	30
Web UI	Account -> SIP -> Advanced Settings -> Keep Alive Interval
Phone UI	Not Available

   Auto Provision & Phone UI: (H3G/H6/Myriad series)

Parameter	AccountXKeepAliveInterval
Description	It configures the keep alive timer For H3G: X=1-3 For H6: X=1-4 For M3/M5/M7: X=1-8 For M8: X=1-20

Permitted Values	[0,*]
Default	40
Web UI	Account -> Advanced -> Keep Alive Timer
Phone UI	Not Available

3. Set the “Register Expire Time” parameter to a shorter or longer time on phone web UI:

Web UI: (H2P)

Account -> SIP -> Register Settings -> SIP Sever 1 -> Registration Expiration

Web UI: (H3G/H6/Myriad series)

Account -> Basic -> Register Expire Time

Auto Provision & Phone UI: (H2P)

Parameter	sip.lineX.RegisterTTL
Description	It configures the interval (in seconds) for the IP phone to retry to re-register account when registration fails, X = 1-2
Permitted Values	[1,65535]
Default	3600
Web UI	Account -> SIP -> Register Settings -> SIP Server 1 -> Registration Expiration
Phone UI	Not Available

Auto Provision & Phone UI: (H3G/H6/Myriad series)

Parameter	AccountXServer1Expire
Description	It configures the registration expiration time (in seconds) of SIP server for accountX. For H3G: X=1-3 For H6: X=1-4 For M3/M5/M7: X=1-8 For M8: X=1-20
Permitted Values	[60,*]
Default	3600
Web UI	Account -> Basic -> Register Expire Time
Phone UI	Not Available

Note Please check with your service provider to see if they have limitation on this as for some PBX, send Register message too frequently will cause the PBX forbidden the account.

6.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

6.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

6.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

7 How to adjust default SIP account

7.1 Issue Summary

Customers may register more than one SIP account in one device. Generally, account 1 is the default account. But sometimes the user needs to switch to the default account so that user can make calls easily. This FAQ will show you how to adjust the default SIP account.

7.2 Possible Causes

NA

7.3 How to Resolve

You can change the default SIP account through the web UI/phone UI or through auto provision of ALE SIP DeskPhones, details below:

H2P:

Phone UI: Not Supported

Web UI:

Features -> Basic Settings -> Enable Default Line

Features -> Basic Settings -> Default Ext Line

H3G/H6/Myriad series:

Phone UI: Menu -> Features -> Deafult Account

Web UI:

Features -> SIP -> Default Account

Auto Provision & Phone UI: (H2P)

Parameter	call.port1.EnableDefLine
Description	It configures whether to enable the default account feature or not.
Permitted Values	0 - Disable. 1 - Enable.
Default	1
Web UI	Features -> Basic Settings -> Enable Default Line
Phone UI	Not Available
Parameter	**call.port1.DefaultExtLine**
Description	It configures the default account.

Permitted Values	1 – Account 1 2 – Account 2
Default	1
Web UI	Features -> Basic Settings -> Default Ext Line
Phone UI	Not Available

Auto Provision & Phone UI: (H3G/H6/Myriad series)

Parameter	**SIPDefaultAccount**
Description	It configures the SIP DeskPhone default account.
Permitted Values	1 - Account 1 2 - Account 2 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
Phone UI	Menu -> Features -> Default Account

7.4 More

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 8.4 Default Account**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

7.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

7.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

8 Phone can't receive any incoming calls

8.1 Issue Summary

Customers may meet the problem that the SIP account is registered, phone can call out normally but can't receive any incoming calls.

8.2 Possible Causes

Phone enabled DND feature may cause this issue.

8.3 How to Resolve

1. Check the phone side to see if there is a DND icon on the LCD screen, if yes, just press the DND softkey to disable this feature will solve this issue.

2. If there is no DND icon on the LCD screen, please check with your service provider to see if the DND feature of this SIP account is enabled on the server side and ask them to disable it to solve this issue.

8.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

8.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

8.6 Firmware Version

All version

9 Phone can't make outgoing call

9.1 Issue Summary

This FAQ shares different reasons cause the situation that phone could not make outgoing call and the corresponding solutions.

9.2 Possible Causes

1. SIP account not configured or registered failed
2. Network issue
3. SIP server limitation

9.3 How to Resolve

1. Please make sure that your SIP account is registered, if not, please refer to FAQ "SIP account register failed"

2. If the account is registered fine, please check with your IT manager to see if there is any problem with the network during your call out.

3. Please check with your service manager to see if any additional prefix needed to meet the SIP trunk rules when calling out, for example:

Add 9 before the number when calling external numbers

Add 8 before the extension when call xx department

9.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

9.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

9.6 Firmware Version

All version

10 How to set up a local 3-way conference call

10.1 Issue Summary

This FAQ shared the quick steps to set up a local 3-way conference call with ALE SIP DeskPhones.

10.2 Possible Causes

NA

10.3 How to Resolve

1. Device information:

Phone A with SIP account A

Phone B with SIP account B

Phone C with SIP account C

Phone A will set up the local 3-way conference call

2. Steps:

A calls B, B answers, A and B establish a call

A press “Conf” key

A dials C, C answers, A and C establish a call

A press the “Conf” key again to establish a local 3-way conference call

The same steps for all other parties.

10.4 More

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 12.2 X-party Conference**) for more information.
If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

10.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

10.6 Firmware Version

All version

11 How to transfer a call with ALE SIP DeskPhones

11.1 Issue Summary

All ALE SIP DeskPhones support call transfer using the REFER method specified in RFC 3515 and offer three types of transfer:

Blind Transfer -- Transfer a call directly to another party without consulting. Blind transfer is implemented by a simple REFER method without Replaces in the Refer-To header.

Attended Transfer (Consultative Transfer) -- Transfer a call with prior consulting. Attended transfer is implemented by a REFER method with Replaces in the Refer-To header.

Semi-attended Transfer (Semi-consultative Transfer) -- Transfer a call when the third party is ringing. Semi-attended transfer is implemented by a REFER method with Replaces in the Refer-To header.

This FAQ shares the detailed steps of these 3 kinds of transfer mode via ALE SIP DeskPhones.

11.2 Possible Causes

NA

11.3 How to Resolve

H2P:

1. Blind Transfer call:

A and B establish a call

A press “XFER” key and input the number of C

A press “XFER” key again to finish the blind transfer process

2. Consultative Transfer call:

A and B establish a call

A press “XFER” key and input the number of C

A press “OK” key or “Dial” or “#” key

A and C establish a call

A press “XFER” key to finish the consultative transfer process

3. Semi-consultative Transfer call:

A and B establish a call

A press “XFER” key and input the number of C

A press “OK” key or “Dial” or “#” key

C rings but not answer the call, A hear the ring back

A press “XFER” key to finish the semi-consultative transfer process

H3G/H6/Myriad series:

1. Blind Transfer call:

A and B establish a call

A press “Transfer” key and input the number of C

A press “B Trsf” key to finish the blind transfer process

2. Consultative Transfer call:

A and B establish a call

A press “Transfer” key and input the number of C

A press “OK” key or “Call” or “#” key

A and C establish a call

A press “Transfer” key to finish the consultative transfer process

3. Semi-consultative Transfer call:

A and B establish a call

A press “Transfer” key and input the number of C

A press “OK” key or “Call” or “#” key

C rings but not answer the call, A hear the ring back

A press “Transfer” key to finish the semi-consultative transfer process

11.4 More

There is some PBX that does not support Blind transfer, if you transfer the call failed, it is advised to check with your service provider first to see if they support it or not.

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 10.19 Call Transfer**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

11.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

11.6 Firmware Version

All version

12 How to set up local forward features with ALE SIP DeskPhones

12.1 Issue Summary

All ALE SIP DeskPhones support the call forward feature which can help customers to forward an incoming call to the target number whenever the conditions are met. There are 3 kind of mode ALE SIP DeskPhones support:

Always Forward (Immediate)

Busy Forward (Busy)

No Answer Forward (No Reply)

This FAQ shares the detailed steps to configure these 3 kinds of forward via ALE SIP DeskPhones.

12.2 Possible Causes

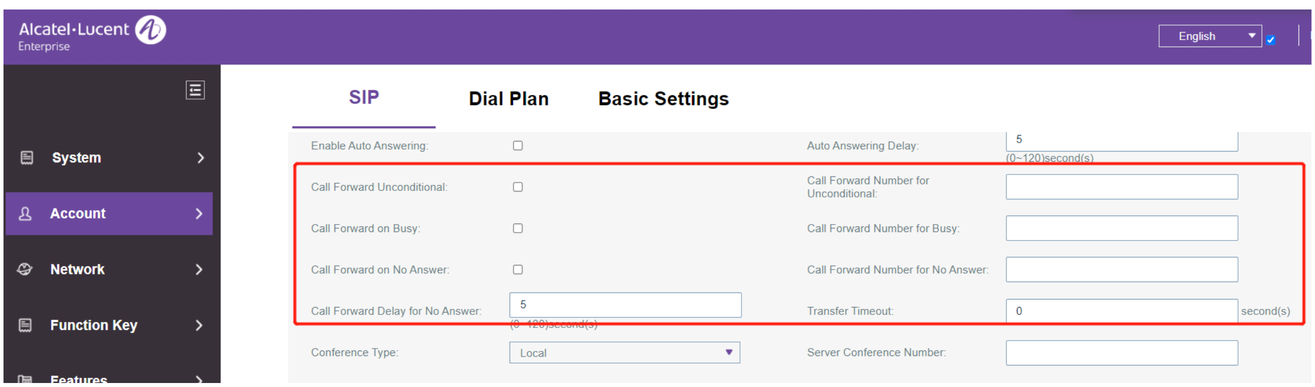
NA

12.3 How to Resolve

H2P:

Phone UI: Not supported

Web UI: Account -> SIP -> Basic Settings



H3G/H6/Myriad series:

Phone UI:

Menu -> Features -> Call Forward

Always Forward: Switch to “Enabled” and fill in the “Forward To” number

Busy Forward: Switch to “Enabled” and fill in the “Forward To” number

No Answer Forward: Switch to “Enabled” and fill in the “Forward To” number

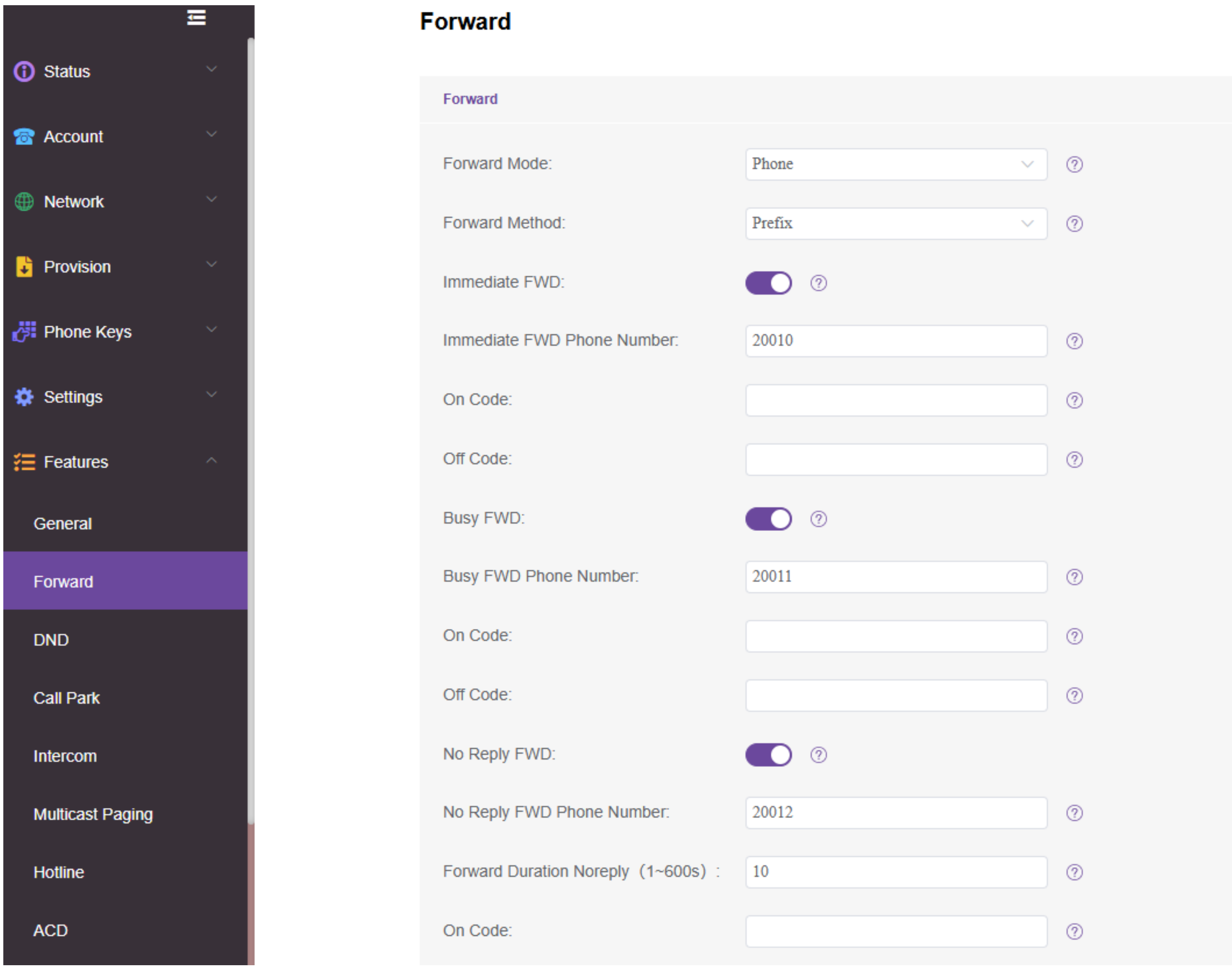
Web UI:

Features -> Forward, select “Forward Mode” to “Phone”

Immediate FWD: Switch to “Enabled” and fill in the “Immediate FWD Phone Number” for always forward

Busy FWD: Switch to “Enabled” and fill in the “Busy FWD Phone Number” for busy forward

No Reply FWD: Switch to “Enabled” and fill in the “No Reply FWD Phone Number” for no answer forward



12.4 More

You can also refer to the [Administration Manual for ALE Myriad and Halo Series DeskPhone](#) (**Chapter 10.14 Call Forward**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

12.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

12.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: All version

13 How to check the status of ALE SIP DeskPhones

13.1 Issue Summary

Some customers need to know how to quickly get the phone's basic information like MAC address, IP address, version information etc. This FAQ will help you.

13.2 Possible Causes

NA

13.3 How to Resolve

H2P:

Press the "OK" key will get the phone status information, detailed below:

Number (Account)
Mode (Network mode)
IPv4 (IP status)
VLAN ID

Note For the MAC/Hardware/Software version, please go to Setting -> Version

H3G/H6/Myriad series:

Press the "OK" key will get the phone status information, detailed below:

IPv4 address
MAC
Wi-Fi MAC (Only for M8)
Version
Network Status (In "More -> Network")
Soft infos (In "More -> Phone")
Hard infos (In "More -> Phone")
Account Status (In "More -> Accounts")

13.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

13.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

13.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

14 How to upgrade ALE SIP DeskPhone binary

14.1 Issue Summary

This FAQ shows how to upgrade ALE SIP DeskPhone binary

14.2 Possible Causes

NA

14.3 How to Resolve

There are three methods to upgrade the ALE SIP DeskPhone binary:

1. Through phone web UI

1) Download the binary that you will use to upgrade

2) Log into the phone web UI by typing the IP address of the phone on the address bar with below format:

H2P: “https://phone IP/” like https://10.10.1.1/

H3G/H6/Myriad series: “https://phone IP/” like https://10.10.1.1/

The default username is “admin” while the password is “123456”

3) Go to the path:

H2P: System -> Upgrade -> Software upgrade

H3G/H6/Myriad series: Maintenance -> Firmware Upgrade -> Upload Firmware(sip*)

4) Click “Select” to select the binary file

Note

For H2P, please select the downloaded “H2P-xxxxxx.z” file directly.

For H3G/H6/Myriad series, please unfold the downloaded file and select the “sip*” file, for example, select the “sipM8” for M8 model, for detailed file name, please refer to the “More” chapter of this FAQ.

5) Click “Update” to start the upgrade process for H3G/H6/Myriad series DeskPhones; Click “Upgrade” to start the upgrade process for H2P DeskPhones.

Note

Do not refresh the page or close the browser, otherwise the upgrade will fail
Do not unplug the network cables and power cables when the IP phone is upgrading firmware

2. Through auto provision

You can use this parameter in your provision template to trigger the phone to upgrade the binary:

H2P:

Parameter	FirmwareUrl
Description	It configures the access URL of the firmware file. Example: <sysConf> <ota> <FirmwareUrl>http://172.16.6.70:8000/x6.z</FirmwareUrl> </ota> </sysConf>
Permitted Values	URL
Default	blank

H3G/H6/Myriad series:

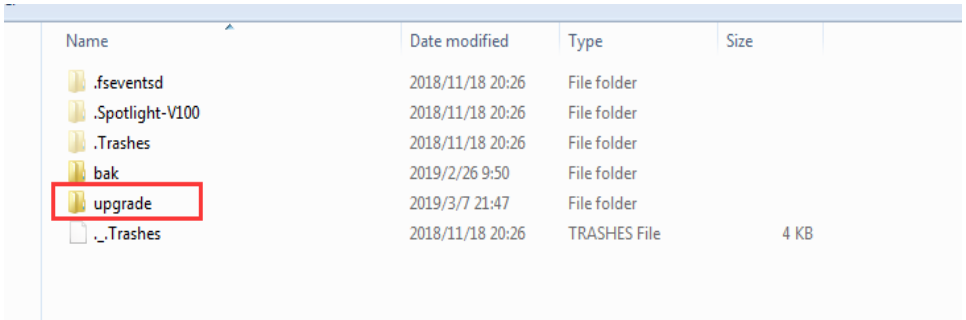
Parameter	DeviceFirmwareUpgradeUrl
Description	It configures the access URL of the firmware file. Example: <setting id="DeviceFirmwareUpgradeUrl" value="http://135.251.222.94:8090/" override="true"/>
Permitted Values	URL within 511 characters
Default	blank

3. USB Upgrade

1) Prepare a moveable USB disk with FAT32 format

2) Create a folder and name it “upgrade”

3) Copy the extracted sip* file to this upgrade folder, for detailed file name, please refer to the “More” chapter of this FAQ.



4) Plug U disk into the phone's USB port

5) Power on the phone

6) During step 1 of initialization process, pressing “4” + “7” + “8” + “*” keys in sequence. Release all keys until all the LEDs are lighted on.

7) Phone will reboot and enter upgrading process.

Note H2P/H3G DeskPhones do not support this method.

14.4 More

You can download the latest firmware online at http://www.aledevice.com/site/download

The following table lists the associated and latest firmware name for each IP DeskPhone model.

Model	Firmware Name
H2P	H2P-xxxxxx.z
H3G/H6	sipH3_6X
M3/M5/M7	sip9000N
M8	sipM8

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 4 Firmware Upgrade**) for more information.

14.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

14.6 Firmware Version

H2P: All version

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

15 What is the pinout of ALE SIP DeskPhone headset port

15.1 Issue Summary

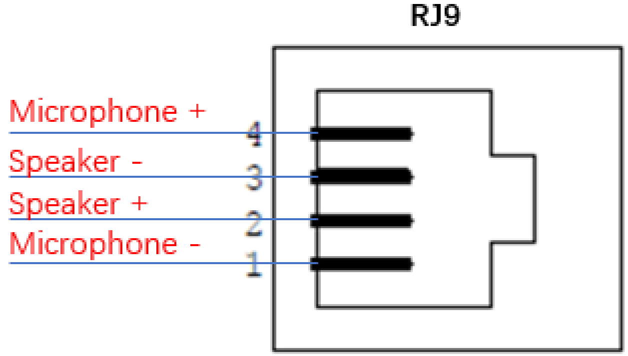
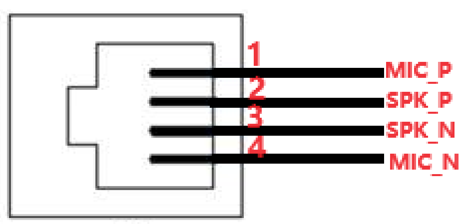
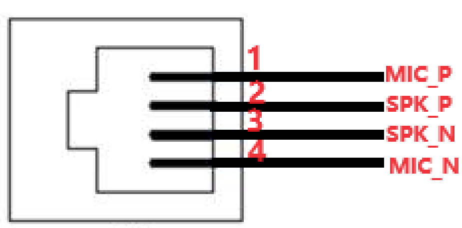
All ALE SIP DeskPhones support wired headsets, and different customers may use different headsets, so the pinout type of headset port is the needed info for the customer to choose the suitable headset.

15.2 Possible Causes

NA

15.3 How to Resolve

The pinout of different phones is different, details below:

Model	Headset Port & Its Pinout
H2P	
H3G/H6	
M3/M5/M7	USB A/C
M8	USB A/C, Bluetooth 5.0 

15.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

15.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

15.6 Firmware Version

All version

16 How to set up the voice mail feature of ALE SIP DeskPhones

16.1 Issue Summary

All ALE IP DeskPhones support voice mail feature which can help caller to send messages when callee is not available. This FAQ shares the detailed steps to set up the voice mail feature of ALE SIP DeskPhones.

16.2 Possible Causes

NA

16.3 How to Resolve

H2P:

Web UI: Account -> SIP -> Basic Settings -> Voice Message Number

Phone UI: Press the Voicemail hard key -> Select Line, press OK -> Voice Mail (Enabled) -> Number

H3G/H6/Myriad series:

Web UI: Account -> Advanced -> Voice Mail Number:

Phone UI: Menu -> Message -> Voicemail -> Set Voicemail Number

You can also dial the Voice Mail Number directly to access the voice mail.

16.4 More

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 12.6 Voicemail**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

16.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

16.6 Firmware Version

H2P: All version

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

17 How to add local phone book contact of ALE SIP DeskPhones

17.1 Issue Summary

This FAQ shares the detailed steps to add contacts to the local phone book of ALE SIP DeskPhones.

17.2 Possible Causes

NA

17.3 How to Resolve

H2P:

Web UI: Phonebook -> Contacts -> Contact List -> Add new contact, fill in the corresponding items needed

Phone UI: More -> Contacts -> All Contacts -> OK -> Add, fill in the corresponding items needed

Note

please login the phone web UI first and go to Function Key -> Softkey -> Softkey Settings, then add the “Local Contacts” to the “Selected Softkeys” when “Screen” is set to “Desktop”.

H3G/H6/Myriad series:

Web UI: Contact Manager -> Local Directory -> Add, fill in the corresponding items needed

Phone UI: Menu -> Directory -> Local Directory -> Add, fill in the corresponding items needed

17.4 More

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 6.1 Local Directory**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

17.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

17.6 Firmware Version

All version

18 How to set up the correct time for your ALE SIP DeskPhones

18.1 Issue Summary

Time & Date is the normal setting that a customer may use when setting up a phone. This FAQ shares the steps to configure it in ALE SIP DeskPhones.

18.2 Possible Causes

NA

18.3 How to Resolve

ALE SIP DeskPhones maintain a local clock. You can choose to get the time and date from SNTP (Simple Network Time Protocol) time server to have the most accurate time and set DST (Daylight Saving Time) to make better use of daylight and to conserve energy, or you can set the time and date manually. The time and date can be displayed in several formats on the idle screen. The detailed settings are shown below:

H2P:

Web UI: Settings -> Time/Date, all settings here, just modify the one you needed

Phone UI: Not supported

H3G/H6/Myriad series:

Web UI:

Setting -> Time&Date, fill in the corresponding items needed

Phone UI:

Menu -> Basic Setting -> Time & Date -> Time & Date Format (here you can change the format of the “Date (YYY-MM-DD.etc.)” & “Time (24 Hour/12 Hour)”).

Menu -> Basic Setting -> Time & Date -> Time & Date Format -> General -> SNTP Settings (here you can change the SNTP used).

Menu -> Basic Setting -> Time & Date -> Time & Date Format -> General -> Manual Settings (here you can manually set the time and date of the phone).

Note For manual time, once the phone is rebooted, the manual set time will be reset to the original one

18.4 More

You can also refer to the [Administration Manual for ALE Myriad and Halo Series DeskPhone] (**Chapter 11.6 Time and Date**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

18.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

18.6 Firmware Version

H2P: All version

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

19 How to reset the administrator password

19.1 Issue Summary

Administrator password is used to login phone web UI or the advanced settings on phone LCD side, but for some management requirements, this password is just for the phone manager, what can we do if the admin password has been changed or forgotten? This FAQ shares the steps to configure it in ALE SIP DeskPhones.

19.2 Possible Causes

1. Customers changed the password but forgot it
2. Administrator changed but password is missing
3. The default password of this version is not “123456”

19.3 How to Resolve

1. Check with your phone provider to see if there is any default password changed.

2. If there is no special default password, please try to factory reset the phones to get the default password:

H2P:

You can long press the “OK” hard key for 8 seconds, then press “OK” to reset the phone to factory settings

H3G/H6/M3/M5/M7:

You can long press the “Conference” hard key for 10 seconds, then press “OK” to reset the phone to factory settings.

M8:

You can long press the “Headset” hard key for 10 seconds, then press “OK” to reset the phone to factory settings.

Note Reset to Factory will clear all the configuration of the phone, if you don't know how to configure it, please kindly contact your phone provider for help first.

19.4 More

You can also refer to the [Administration Manual for ALE Myriad and Halo Series DeskPhone] (**Chapter 13.2 Resetting Device to Factory Settings**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

19.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

19.6 Firmware Version

All Version

20 How to change the phone local ringtone

20.1 Issue Summary

For the ringing for incoming calls, different customers may like different rings, normally it is a personal setting, and sometimes, customers may have questions below:

1. Phone doesn't ring the correct ring tone you set locally
2. There is some external Beep before the phone rings, but you don't want that
3. How to make the phone rings with headset when you use a headset

This FAQ shares the method to adjust the local incoming ring tone through ALE SIP DeskPhone LCD side including below options:

1. Adjust ring tones
2. Adjust ring mode
3. Adjust Beep options
4. Adjust ring device

20.2 Possible Causes

1. Ring tone and ring mode being modified
2. The incoming call matches the distinctive ring tones
3. Beep options being changed

20.3 How to Resolve

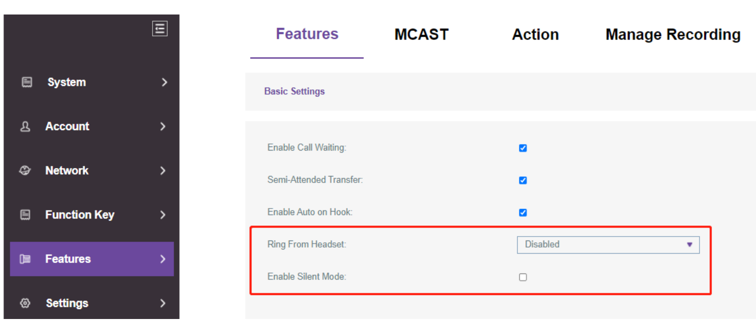
H2P:

1. Press "Setting" -> "Phone" -> "Ringing" -> "Ring Type"

2. Adjust to select the ring tone you want and press "OK" key to save it

3. For "Ring mode" & "Beep", H2P DeskPhones don't support for now

4. For "Ring Device" & "Silent Mode", you can configure them through web UI: "Features" -> "Basic Settings"



H3G/H6/Myriad series:

1. Press “Menu” -> “Basic Setting” -> “Sound” -> “Ringing”

2. Press “Int Melody” key to configure the internal calls, the supported ringtone will list, select it and press “OK” key to save the configuration.

3. Press “Ext Melody” key to configure the External calls from server side, the supported ringtone will list, select it and press “OK” key to save the configuration.

4. Press “Ring Mode” key to configure the different ringing mode you want:

Normal ringing: phone will ring with a fixed volume

Progressive: phone will ring with a dynamic volume from low to loud

Silent mode: Phone will ring with no voice

5. Press “Beep” key to configure the different beep options you want:

0 beep / 1 beep / 3 beeps

6. Press “Ring Device” key to configure the different ring device when you enabled the headset mode:

Handsfree: In headset mode, phone will ring with the speaker

Headset: In headset mode, phone will ring with the headset

HF+HE: In headset mode, phone will ring with the speaker & headset at the same time

Note The ring tone priority from high to low: distinctive ring tones -> local ring tones

20.4 More

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 7.3 Ring Tones**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

20.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

20.6 Firmware Version

All Version

21 How to change the phone LCD language

21.1 Issue Summary

Different customers may like to use different languages on phone LCD side, this FAQ shares the method to adjust the phone LCD Language quickly.

21.2 Possible Causes

Custom language requirement.

21.3 How to Resolve

H2P:

1. Press “Setting” -> “Phone” -> “Language”

2. The supported language will list, select the one needed and press “OK” key to save the configuration

H3G/H6/Myriad series:

1. Press “Menu” -> “Basic Setting” -> “Language”

2. The supported language will list, select the one needed and press “OK” key to save the configuration.

21.4 More

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 11.1 Multiple Languages**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

21.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

21.6 Firmware Version

All Version

22 How to configure the programmable key of ALE SIP DeskPhone

22.1 Issue Summary

Users can customize programmable keys on the phone to access usually used functions like BLF, Speed dial etc. This FAQ shares the method to quickly set up a programmable key with Speed dial and BLF.

22.2 Possible Causes

NA

22.3 How to Resolve

H2P:

1. There are totally 2 keys can be programable set, just select one key and long press it for 2 seconds, then you will see the Type option, default value is "Line"

2. Press the Right or Left navigation key to switch the Type to "Memory Key"

3. Press the Down navigation key, you will see the Subtype option.

4. Press the Right or Left navigation key to switch the Type to "Speed Dial" or "BLF/New Call"

5. When the key Subtype is changed to "Speed Dial", the corresponding information needed will list automatically, the "Tel" is the number you need to do speed dial.

6. When the key type is changed to "BLF/New call", the corresponding information needed will list automatically, the "Tel" is the number you need to monitor, the "Pickup Number" is the pickup code used to pick up a call if needed.

7. Press "OK" key to save the configuration

H3G/H6/Myriad series:

1. Select one key and long press it for 2 seconds, then you will see the Key Type option, default value is "Undefined"

2. Press the Right or Left navigation key to switch the Key Type to the one you need like "Speed Dial" or "BLF"

3. When the key type is changed to "Speed Dial", the corresponding information needed will list automatically, the "Value" is the number you need to do speed dial.

4. When the key type is changed to "BLF", the corresponding information needed will list automatically, the "Value" is the number you need to monitor, the "Extension" is the pickup code used to pick up a call if needed.

5. Press “OK” key to save the configuration.

22.4 More

ALE SIP DeskPhones support programmable key numbers and types (key types may be different according to different firmware version):

Model	Programmable Keys
H2P	2
H3G	8
H6	12
M3	20
M5	28
M7	28
M8	36

Model	Key Types	Key Subtypes
H2P	None	
	Memory Key	None, BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, BLF/DTMF, Presence, MWI, Speed Dial, Intercom, Call park, Call forward
	Line	
	Key Event	None, MWI, Do-not-disturb, Call hold, Call transfer, Phonebook, Redial, Pickup, Join, Call forward, Call Logs, Flash, Memo, Headset, Release, Lock phone, Call Back, Hide DTMF, Intercom, Group Listening, Prefix, Hot Desking, Agent, End, Disposition, Escalate, Trace, Handfree, Answer Key, Private Hold, Local Contacts, LDAP Group, XML Group, Broadsoft Group, Record, Auto Headset
	DTMF	
	URL	
	BLF List Key	
	MCAST Paging	
	Action URL	
	XML Browser	
	MCAST Listening	

Model	Key Types
H3G/H6/Myriad series	N/A
	Speed Dial
	BLF List
	Do Not Disturb
	Directory
	Voice Mail
	Conference
	Forward
	Transfer
	Group Listening
	Headset
	Hot Desking
	Phone Lock
	Prefix
	DTMF
	Direct Pickup
	Group Pickup
	Call Park
	Recall
	XML Browser
	Intercom
	Retrieve Park
	Audio Hub
	Private Hold
	Hold
	BLF
	Account
	USB Recording

You can also refer to the Administration Manual for ALE Myriad and Halo Series DeskPhone (**Chapter 11.10 Programmable Keys**) for more information.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

22.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

22.6 Firmware Version

H2P: 2.10.000.0001083 and above

H3G/H6/Myriad series: 2.14.17.xxx.xxxx and above

23 Tested Wi-Fi Dongle List for ALE SIP DeskPhones

23.1 Issue Summary

This FAQ shares the tested Wi-Fi dongle list for ALE SIP DeskPhones

23.2 Possible Causes

NA

23.3 How to Resolve

Tested Wi-Fi Dongle	Supported Model
Tenda U3	H6/M3/M5/M7
Tenda U9	H6/M3/M5/M7
TP-Link TL-WN725N	H6/M3/M5/M7

23.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

23.5 Supported Models

H6/M3/M5/M7.

23.6 Firmware Version

H6 (Tenda U3/U9): All version

H6 (TP-Link TL-WN725N): 2.14.18.000.2541

M3/M5/M7 (Tenda U3/U9): 2.11.01.1602 and above

M3/M5/M7 (TP-Link TL-WN725N): 2.14.04.000.2359 and above

24 How to debug ALE H3G/H6/Myriad series DeskPhones – Basic

24.1 Issue Summary

Normally, if any issue that needs ALE analysis, provide all of the below debug files at one time will help to locate and solve the issue faster.

1. Pcap file（.pcap）:

Record the signaling between the phone and the server as well as network and voice-related information in the entire call process for troubleshooting signaling issues.

2. Debug log file（.tgz） :

Record the corresponding log information generated by related operations on the phone to troubleshoot problems with the phone itself.

3. Phone configuration file（.xml） :

Phone configuration information, used to troubleshoot whether there is an incorrect configuration,

Debug files which contain your privacy will be kept within ALE and only used for debugging purposes.

24.2 Possible Causes

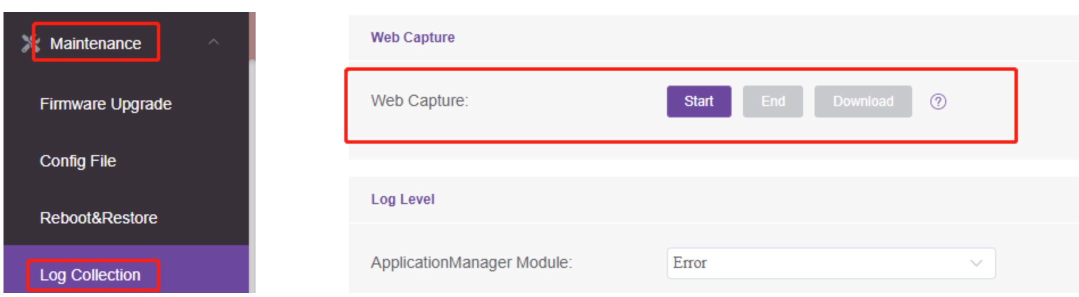
NA

24.3 How to Resolve

Capture pcap file:

Web UI: Maintenance -> Log Collection -> Web Capture

1. Click “Start” to start capturing pcap file
2. Reproduce the issue
3. Click “End” to end capturing pcap file
4. Click “Download” to export and save the pcap file



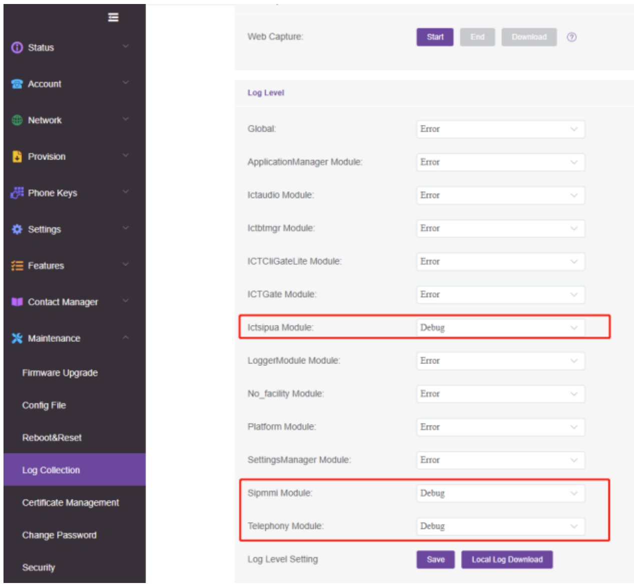
Download local log:

Web UI: Maintenance -> Log Collection -> Log Level

1. Set “Ictsipua Module” & “Sipmmi Module” & “Telephony Module” to “Debug” and save

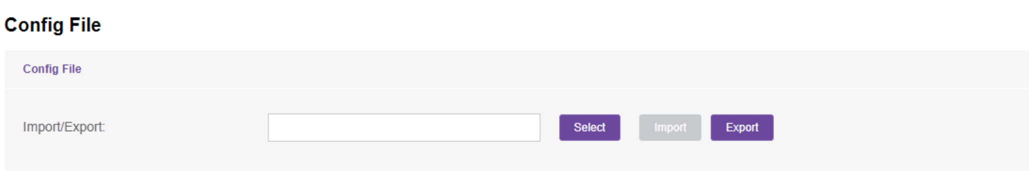
2. Reproduce the issue

3. Click “Local Log Download” to export and save the log file



Export Config File:

Web UI: Maintenance -> Config File, Click “Export” to download and save the configuration file



Note Please export the debug files immediately after testing, just in case of the file being overwritten and the pcap file and debug log file should be exported from the same test so that they can match each other.

24.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

24.5 Supported Models

H3G/H6/M3/M5/M7/M8.

24.6 Firmware Version

H3G/H6: 2.12.xx.xx version and above

M3/M5/M7/M8: 2.13.xx.xx version and above

25 How to debug ALE H3G/H6/Myriad series DeskPhones – Advanced

25.1 Issue Summary

Normally, if any issue that need ALE analysis, provide the pcap file, config file and syslog debug files at one time will help to locate and solve the issue faster, but for some scenario like phone crash or need long time to reproduce this issue, then export files from phone web UI is not enough, you need to use technical tools in order to get them, this FAQ describe how to get them through the tools.
Debug files which contain your privacy will be kept within ALE and only used for debugging purposes.

25.2 Possible Causes

NA

25.3 How to Resolve

1. Pcap file:

Capture packets with Wireshark tool:

Before using Wireshark to capture packets, you need a hub or switch which support mirror mapping, steps:

1. Connect the phone LAN port to the hub or switch through the network cable.
2. Connect the PC to the same hub or switch through the network cable.
3. Start Wireshark and select the "Ethernet" which has data transmission.
4. Click "Start" button, then you will see the data of the phone.

You can use the filter below to see if the data capture is useful or not, normally SIP message is a mandatory requirement.

SIP

It can filter all SIP messages, if you need to filter the SIP message for exact phone,
please use sip&&ip.addr=="phone ip"

RTP

It can filter all call RTP stream

LLDP

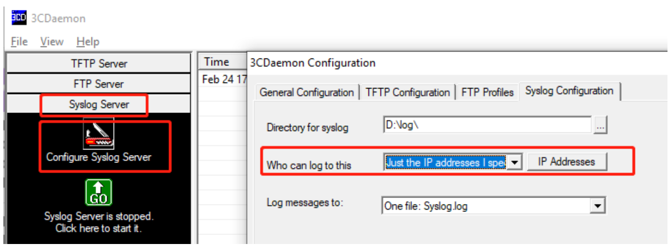
It can filter all LLDP message for VLAN parameters

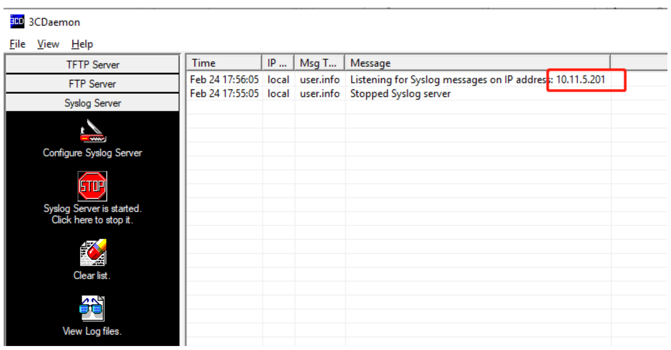
5. Reproduce the issue
6. Click "Stop" to end capturing the pcap file
7. Click "Save" to save the pcap file to your local PC.

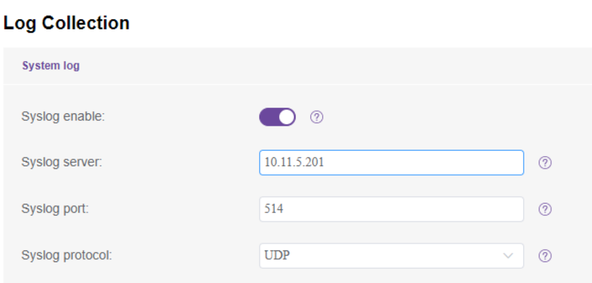
2. Log file:

Export syslog to syslog server (Tool 3CDaemon):

1. Run the tool “3CDaemon”, go to “Syslog Server”
2. Click “Configure Syslog Server”, assign phone IP
3. Click the green “GO” icon to start this server
4. Login phone web UI, go to: “Maintenance -> Log Collection -> System Log”
5. Enable System Log
6. Fill in the syslog server IP, port 514 by default
7. Reproduce the issue
8. Go to the syslog tool directory to get the log file

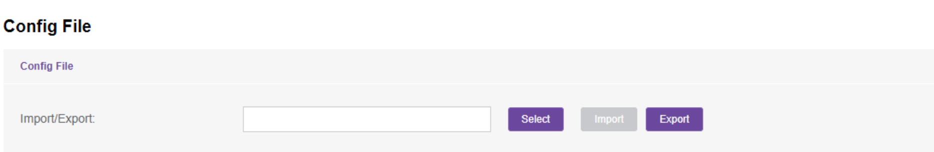






3. Configuration file:

Web UI Path: Maintenance -> Config File, click “Export” to download and save the configuration file



Note

1. Please export the debug files immediately after testing, just in case of the file being overwritten and the pcap file and debug log file should be exported from the same test so that they can match each other.
2. For some scenarios like phone can't get IP address, then it is very complex to login phone web to export configuration file or log file, then just kindly provide the pcap file with the issue description as detailed as

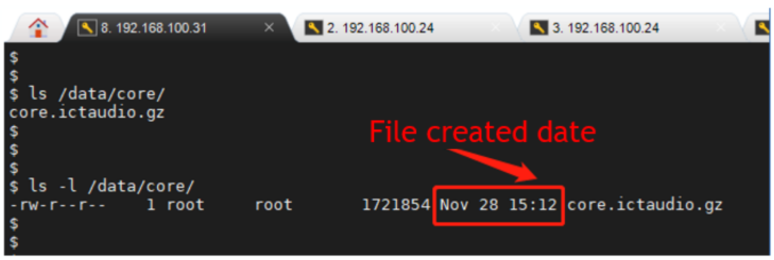
4. Export the core file after phone crash or freeze (Tool MobaXterm):

1. Phone enabled SSH
2. Run tool MobaXterm, select SSH, enter phone IP
3. Login user name password same as the Web UI
4. Use below command to list the core file

   ls /data/core/

5. Use below command to list the detailed info of the core file, make sure the file date is newest

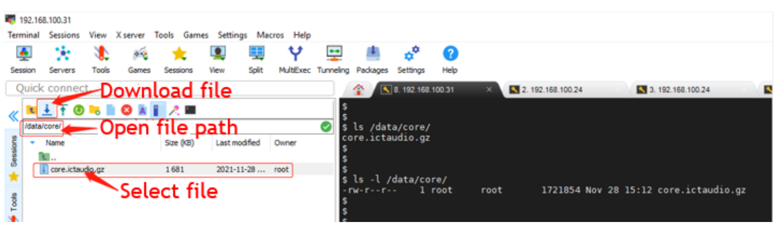
   ls –l /data/core/



6. Open the core file path at the left address bar of the tool

   /data/core/

7. Select the correct core file and click the download button to download the core file



25.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

25.5 Supported Models

H3G/H6/M3/M5/M7/M8.

25.6 Firmware Version

H3G/H6: 2.12.xx.xx version and above

M3/M5/M7/M8: 2.13.xx.xx version and above

26 How to debug ALE SIP DeskPhones – H2P

26.1 Issue Summary

Normally, if any issue that need ALE analysis, provide all below debug files at one time will help to locate and solve the issue faster:

1. Pcap file (.pcap):

Record the signaling between the phone and the server as well as network and voice-related information in the entire call process for troubleshooting signaling issues.

2. Debug log file (.txt or .log):

Record the corresponding log information generated by related operations on the phone to troubleshoot problems with the phone itself.

3. Phone configuration file (.xml):

Phone configuration information, used to troubleshoot whether there is an incorrect configuration.
At the same time, for some scenario like phone crash or need long time to reproduce this issue, then export files from phone web UI is not enough, you need to use technical tools in order to get them, this FAQ describes how to get them through phone web or through tools.
Debug files which contain your privacy will be kept within ALE and only used for debugging purposes.

26.2 Possible Causes

NA

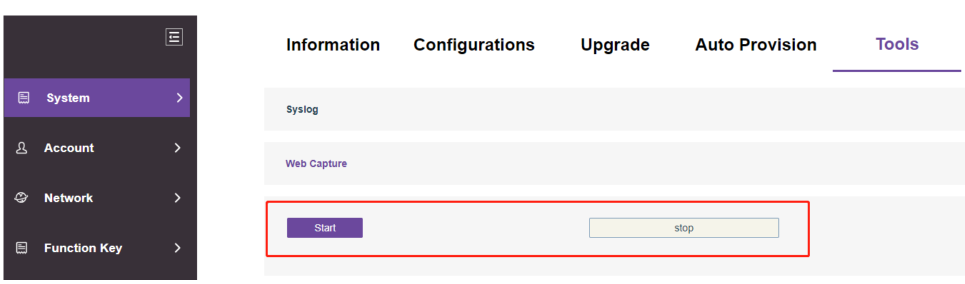
26.3 How to Resolve

1. Pcap file:

Capture packets through web UI:

Web UI Path: System -> Tools -> Web Capture

1. Click "Start" to start capturing pcap file
2. Reproduce the issue
3. Click "Stop" to end capturing and download the pcap file



Capture packets with Wireshark tool:

Before using Wireshark to capture packets, you need a hub or switch which support mirror mapping, steps:

1. Connect the phone LAN port to the hub or switch through the network cable.
2. Connect the PC to the same hub or switch through the network cable.
3. Start Wireshark and select the “Ethernet” which has data transmission.
4. Click “Start” button, then you will see the data of the phone.

You can use the filter below to see if the data capture is useful or not, normally SIP message must need.

SIP

It can filter all SIP messages, if you need to filter the SIP message for exact phone,

please use sip&&ip.addr==“phone ip”

RTP

It can filter all call RTP stream

LLDP

It can filter all LLDP message for VLAN parameters

5. Reproduce the issue
6. Click “Stop” to end capturing the pcap file
7. Click “Save” to save the pcap file to your local PC.

2. Log file:

Download local log through web UI:

Web UI Path: System -> Device Log -> Device Log

1. Click “Start”
2. Reproduce the issue
3. Click “Stop”, then “Save” to export and save the log file

Web UI Path: System -> Tools -> Syslog

Web UI Path: System -> Device Log -> Device Log

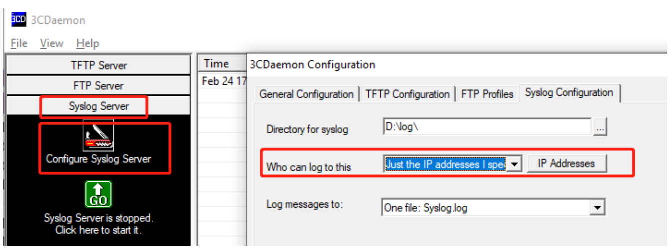
1. Select "Enable Syslog"
2. Server Address keep as 0.0.0.0
3. Server Port keep as 514
4. Select "APP Log Level" to debug
5. Select "Export Log"
6. Apply to save the settings

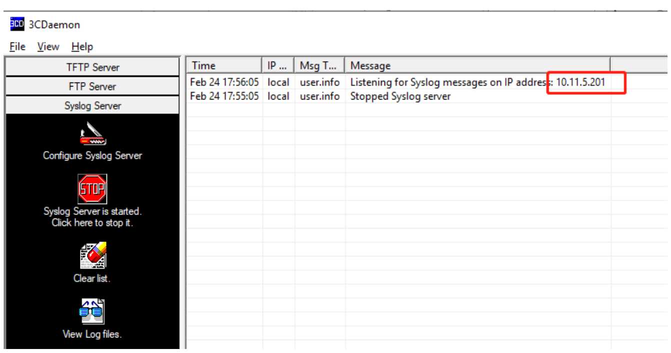
7. Reproduce the issue
8. Go to System -> Tools -> Export Log, click "Export Log" to save the og file

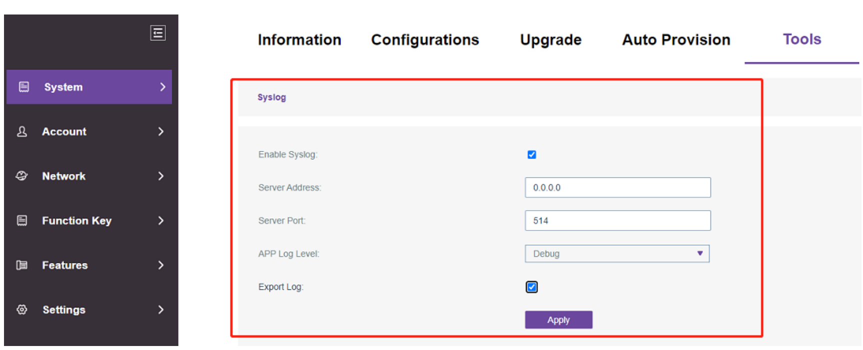
Send the two log files above to ALE

Export syslog to syslog server (Tool 3CDaemon):

1. Run the tool “3CDaemon”, go to “Syslog Server”
2. Click “Configure Syslog Server”, assign phone IP
3. Click the green “GO” icon to start this server
4. Login phone web UI, go to: “System -> Tools -> Syslog”
5. Enable System Log
6. Fill in the syslog server IP, port 514 by default
7. APP log level select to “Debug”
8. Select the “Export Log” also, then click “Apply”
9. Reproduce the issue
10. Go to the syslog tool directory to get the log file

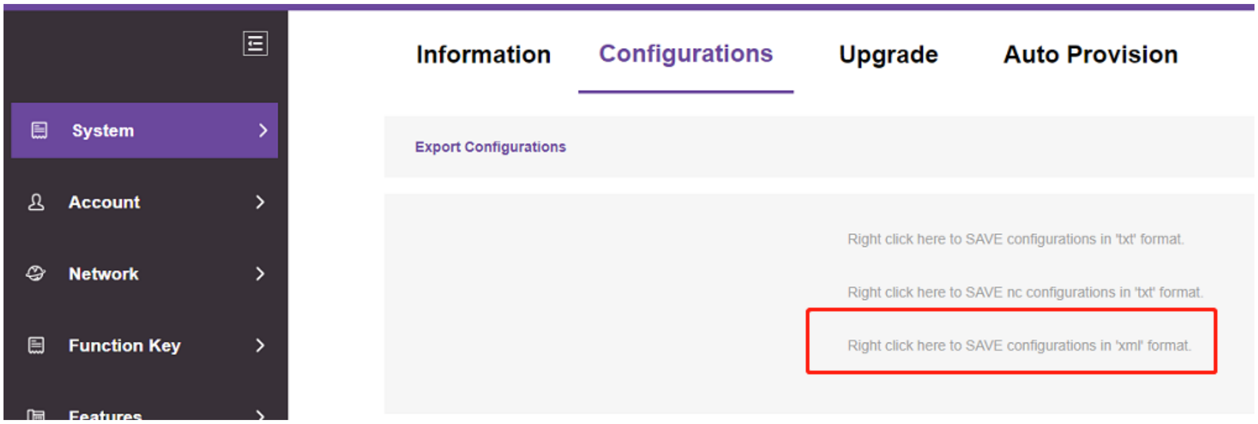






3. Configuration file:

Web UI Path: System -> Configurations -> Export Configurations, it is advised to export the ".xml" format configuration according to the guidance.

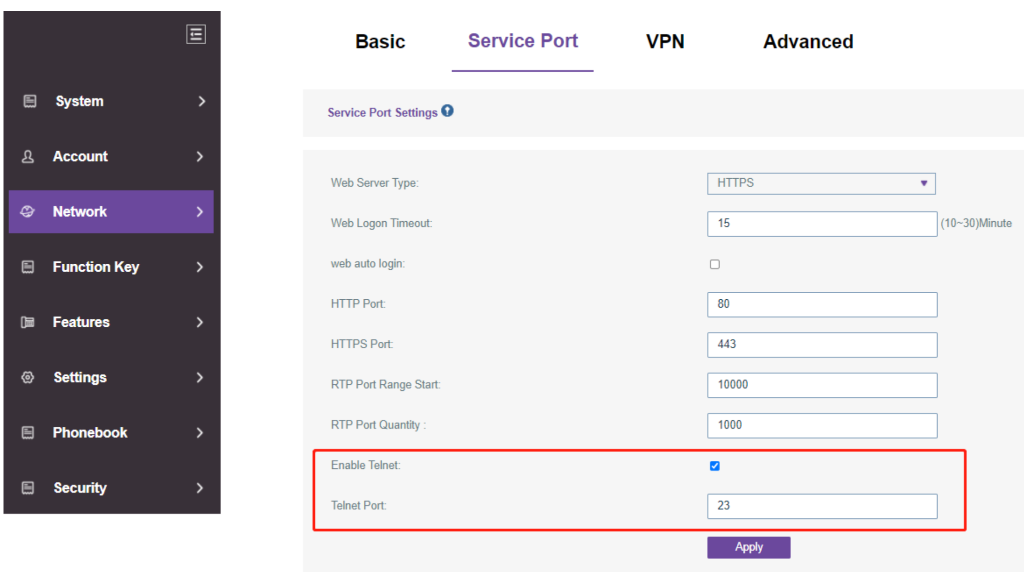


Note

1. Please export the debug files immediately after testing, just in case of the file being overwritten and the pcap file and debug log file should be exported from the same test so that they can match each other.
2. For some scenarios like phone can't get IP address, then it is very complex to login phone web to export configuration file or log file, then just kindly provide the pcap file with the issue description as detailed as possible, thanks.

4. Export the core file after phone crash or freeze (Tool MobaXterm):

1. Go to phone web UI: Network -> Service Port -> Enable Telnet
2. Enable it and set the telnet port to 23



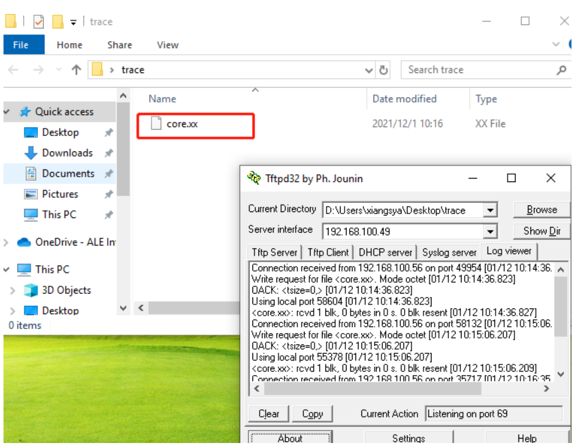
3. Open SSH tool, Click "Session", Click "Telnet", input the phone IP address
4. Login with username "root"
5. Input below command

   cd /mnt/cores/
6. Input below command to show the files start with core

   ls
7. User below command and tftp server to download the core file to your local PC

   tftp -p -r core file name tftp server ip

```
root@dvf97:~# cd /mnt/cores/
root@dvf97:/mnt/cores#
root@dvf97:/mnt/cores# ls
core.xx
root@dvf97:/mnt/cores#
root@dvf97:/mnt/cores#
root@dvf97:/mnt/cores# tftp -p -r core.xx 192.168.100.49
root@dvf97:/mnt/cores#
```



26.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

26.5 Supported Models

H2P

26.6 Firmware Version

All Version

27 How to quickly generate H3G/H6/MX configuration files in batches

27.1 Issue Summary

Customers usually need to deploy a lot of phones at one time with different SIP accounts and it is time-consuming to create auto provision files one by one, this FAQ show in detail about how to quickly generate phone configuration files in batches, this will make the deployment of ALE SIP DeskPhones more convenient and efficient.

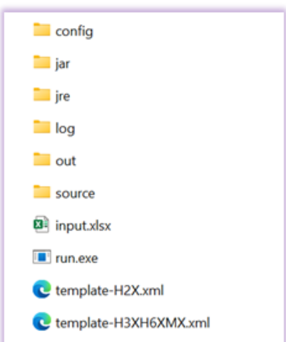
27.2 Possible Causes

NA

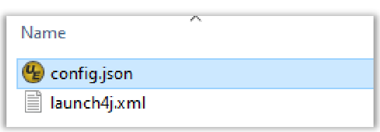
27.3 How to Resolve

ALE provides a tool named "ALE SIP Phone Config File Generation Tool_V1.1" which can be used to create auto provision configuration files in batches, please follow below steps to use it:

1. Unzip the tool



2. Enter the "config" folder, then edit the config.json file.



The format of configuration file is "$$$mac$$$.xml" by default.

This format is for H2P. Please change it to “config.$$$mac$$$” for H3G/H6/M3/M5/M7/M8, where the “$$$mac$$$” is the variable replacement character.

The “$$$mac$$$” will be replaced by the real MAC which is written in the “input.xlsx” file during the generation of configuration files. For example, if the format is “config.$$$mac$$$.xml”, then the configuration file generated will be “config.112233445566.xml” where the “112233445566” is a real MAC of the phone.

3. Open the “input.xlsx” file, then fill in the variable name and its corresponding value for the phones.

(1) Please fill in the variable name in the first row, where the first row and first column must be “MAC Address”. Other parameters, such as: register name, username, password, can be customized and with no ordering limitation. The variable name except “MAC Address” can be English or other languages, as shown below:

A	B	C	D	E	F	G	H	I
MAC Address	IP	Subnet	Router	DNS1	Username	RegisterName	Password	SIPServer
12:23:34:45:56:67	192.168.1.2	255.255.255.0	192.168.1.1	192.168.1.1	1000	1000	qwe123	192.168.1.100
23:34:45:56:67:78	192.168.1.3	255.255.255.0	192.168.1.1	192.168.1.1	1001	1001	test123	192.168.1.100
334455667788	192.168.1.4	255.255.255.0	192.168.1.1	192.168.1.1	1002	1002	sdg123	192.168.1.100

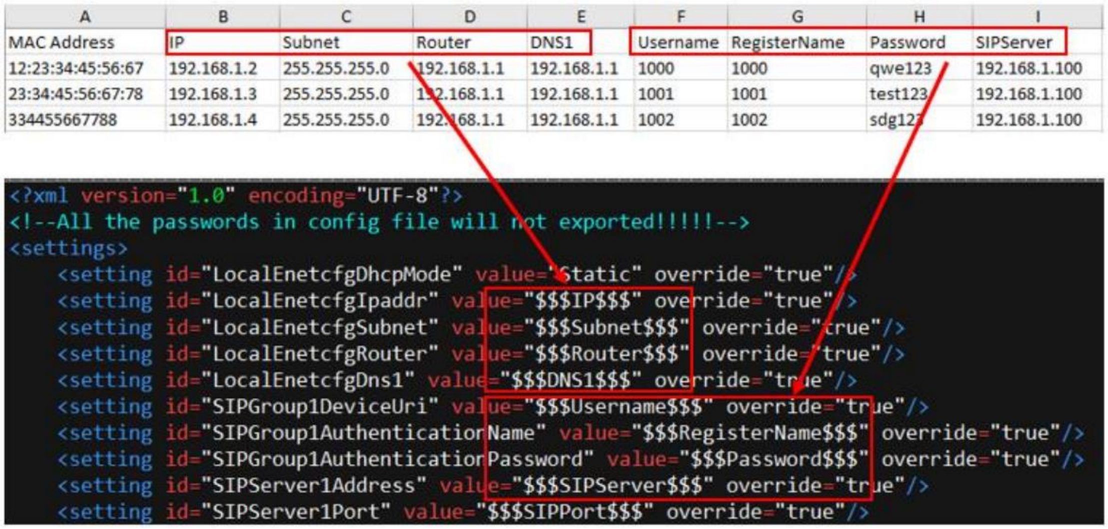
(2) Fill in the corresponding values of the parameters from the second row. The value of MAC Address supports two formats, one with “:” symbol, like “aa:bb:cc:dd:ee:01”, another one without “:” symbol, like “aabbccddee01”.

(3) If the value of a related parameter is empty in the table, then the value of the related parameter in the configuration file generated is also empty. The phone will use the default value of this parameter instead.

4. Rename the “template-H3XH6XMX.xml” to “template.xml”.

Open the template.xml file, this template.xml file includes all of the parameters for H3XH6XMX, user can directly fill in the value of the parameter which is same for all phones, such as SIP Server, NTP Server etc. For the same parameter but with different values for different phones, we need to use the variable symbol to get the values of related parameters from the “input.xlsx” file, such as username, password etc., the configuration rules of variable symbol as shown below.

Open the input.xlsx file, then copy the customized parameters to the value part of the related parameters in template.xml file, the format in template.xml is “$$$customized parameter$$$”, the “$$$customized parameter$$$” will be replaced by the real value from the input.xlsx file during the generation of configuration files, as the picture shown below:



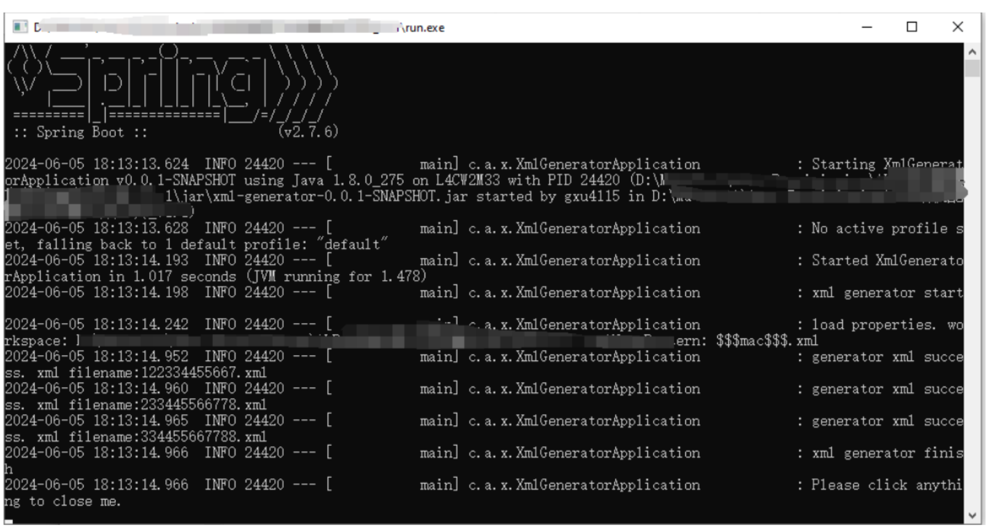
A	B	C	D	E	F	G	H	I
MAC Address	IP	Subnet	Router	DNS1	Username	RegisterName	Password	SIPServer
12:23:34:45:56:67	192.168.1.2	255.255.255.0	192.168.1.1	192.168.1.1	1000	1000	qwe123	192.168.1.100
23:34:45:56:67:78	192.168.1.3	255.255.255.0	192.168.1.1	192.168.1.1	1001	1001	test123	192.168.1.100
334455667788	192.168.1.4	255.255.255.0	192.168.1.1	192.168.1.1	1002	1002	sdg123	192.168.1.100

```xml
<?xml version="1.0" encoding="UTF-8"?>
<!--All the passwords in config file will not exported!!!!!-->
<settings>
    <setting id="LocalEnetcfgDhcpMode" value="Static" override="true"/>
    <setting id="LocalEnetcfgIpaddr" value="$$$IP$$$" override="true"/>
    <setting id="LocalEnetcfgSubnet" value="$$$Subnet$$$" override="true"/>
    <setting id="LocalEnetcfgRouter" value="$$$Router$$$" override="true"/>
    <setting id="LocalEnetcfgDns1" value="$$$DNS1$$$" override="true"/>
    <setting id="SIPGroup1DeviceUri" value="$$$Username$$$" override="true"/>
    <setting id="SIPGroup1AuthenticationName" value="$$$RegisterName$$$" override="true"/>
    <setting id="SIPGroup1AuthenticationPassword" value="$$$Password$$$" override="true"/>
    <setting id="SIPServer1Address" value="$$$SIPServer$$$" override="true"/>
    <setting id="SIPServer1Port" value="$$$SIPPort$$$" override="true"/>
```

The other parameters in the template.xml file of H3XH6XMX are default parameters, there is no need to edit if not necessary.

Note The parameter can be customized, but the parameter name in the template.xml and input.xlsx must be matched for each other.

5. Double click “run.exe” after completing the process above. The configuration files were generated when “Please click anything to close me” is displayed in the bottom line, just click any key to exit this program, as the picture shown below:



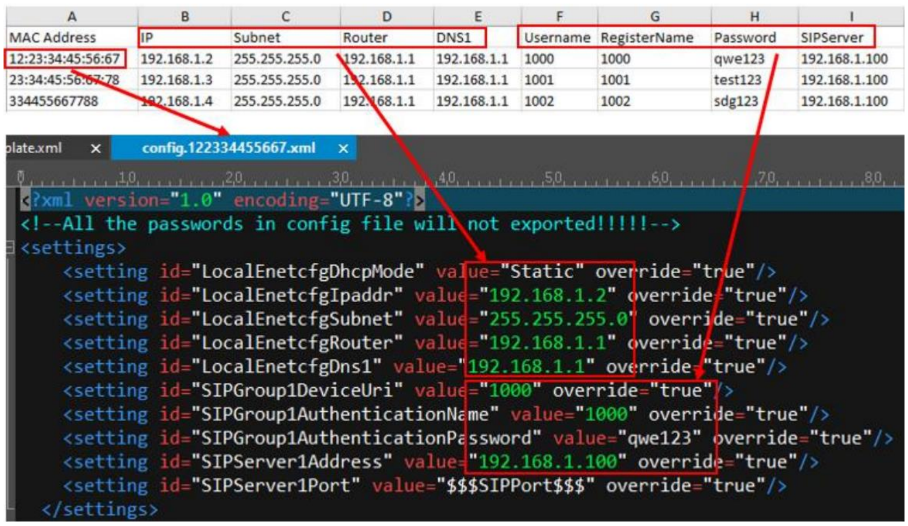
```
D[redacted]\run.exe

:: Spring Boot ::                (v2.7.6)

2024-06-05 18:13:13.624  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : Starting XmlGenerat
orApplication v0.0.1-SNAPSHOT using Java 1.8.0_275 on L4CW2M33 with PID 24420 (D:\[redacted]
[redacted]1\jar\xml-generator-0.0.1-SNAPSHOT.jar started by gxu4115 in D:\[redacted]
2024-06-05 18:13:13.628  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : No active profile s
et, falling back to 1 default profile: "default"
2024-06-05 18:13:14.193  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : Started XmlGenerato
rApplication in 1.017 seconds (JVM running for 1.478)
2024-06-05 18:13:14.198  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : xml generator start

2024-06-05 18:13:14.242  INFO 24420 --- [[redacted]] c.a.x.XmlGeneratorApplication            : load properties. wo
rkspace: [redacted]ern: $$$mac$$$.xml
2024-06-05 18:13:14.952  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : generator xml succe
ss. xml filename:122334455667.xml
2024-06-05 18:13:14.960  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : generator xml succe
ss. xml filename:233445566778.xml
2024-06-05 18:13:14.965  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : generator xml succe
ss. xml filename:334455667788.xml
2024-06-05 18:13:14.966  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : xml generator finis
h
2024-06-05 18:13:14.966  INFO 24420 --- [           main] c.a.x.XmlGeneratorApplication            : Please click anythi
ng to close me.
```

6. Enter the folder “out”, all of the configuration files generated are here. Open any one of the configuration files, then check the value of the parameters are the same as the one defined in the “input.xlsx” file, as the picture shown below:



27.4 More

For other issues related with the mass generation of configuration files, please refer to the ALE SIP DeskPhones Configuration File Mass Generation and Installation Guide.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

27.5 Supported Models

H3G/H6/M3/M5/M7/M8.

27.6 Firmware Version

All Version

28 How to quickly generate H2P DeskPhone configuration files in batches

28.1 Issue Summary

Customers usually need to deploy a lot of phones at one time with different SIP accounts and it is time-consuming to create auto provision files one by one, this FAQ show in detail about how to quickly generate phone configuration files in batches, this will make the deployment of ALE SIP DeskPhones more convenient and efficient.

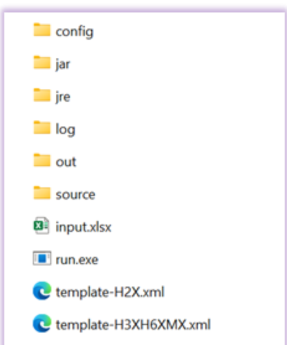
28.2 Possible Causes

NA

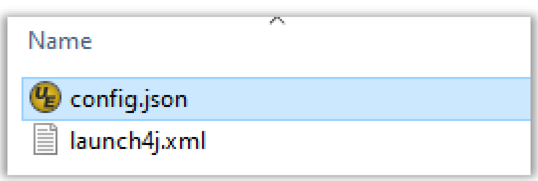
28.3 How to Resolve

ALE provides a tool named "ALE SIP Phone Config File Generation Tool_V1.1" which can be used to create auto provision configuration files in batches, please follow below steps to use it:

1. Unzip the tool



2. Enter the "config" folder, then edit the config.json file.



The format of configuration file is "$$$mac$$$.xml" by default.

The “$$$mac$$$” will be replaced by the real MAC which is written in the “input.xlsx” file during the generation of configuration files. For example, if the format is “$$$mac$$$.xml”, then the configuration file generated will be “112233445566.xml”, where the “112233445566” is a real MAC of the phone.

3. Open the “input.xlsx” file, then fill in the variable name and its corresponding value for the phones.

(1) Please fill in the variable name in the first row, where the first row and first column must be “MAC Address”. Other parameters, such as: register name, username, password, can be customized and with no ordering limitation. The variable name except “MAC Address” can be English or other languages, as shown below:

A	B	C	D	E	F	G	H	I
MAC Address	IP	Subnet	Router	DNS1	Username	RegisterName	Password	SIPServer
12:23:34:45:56:67	192.168.1.2	255.255.255.0	192.168.1.1	192.168.1.1	1000	1000	qwe123	192.168.1.100
23:34:45:56:67:78	192.168.1.3	255.255.255.0	192.168.1.1	192.168.1.1	1001	1001	test123	192.168.1.100
334455667788	192.168.1.4	255.255.255.0	192.168.1.1	192.168.1.1	1002	1002	sdg123	192.168.1.100

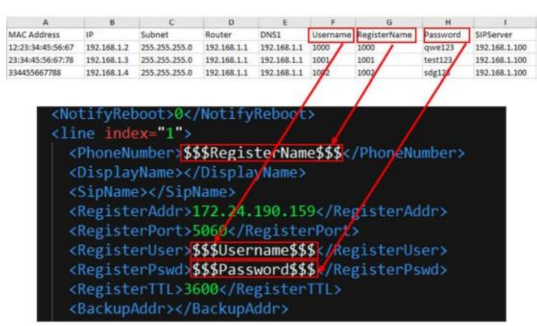
(2) Fill in the corresponding values of the parameters from the second row. The value of MAC Address support two formats, one with “:” symbol, like “aa:bb:cc:dd:ee:01”, another one without “:” symbol, like “aabbccddee01”.

(3) If the value of a related parameter is empty in the table, then the value of the related parameter in the configuration file generated is also empty. The phone will use the default value of this parameter instead.

4. Rename the “template-H2X.xml” to “template.xml”.

Open the template.xml file, this template.xml file includes all of the parameters for H2P, user can directly fill in the value of parameter which is same for all phones, such as SIP Server, NTP Server etc. For the same parameter but with different values for different phones. we need to use the variable symbol to get the values of related parameters from the “input.xlsx” file, such as username, password etc., the configuration rules of variable symbol are shown below.

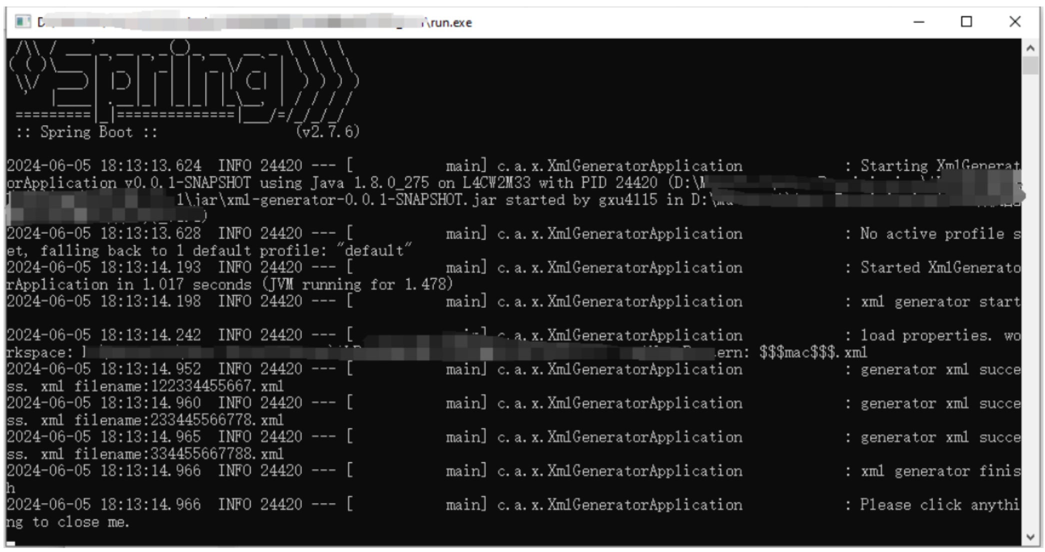
Open the input.xlsx file, then copy the customized parameters to the value part of the related parameters in template.xml file, the format in template.xml is “$$$customized parameter$$$”, the “$$$customized parameter$$$” will be replaced by the real value from the input.xlsx file during the generation of configuration files, as the picture shown below:



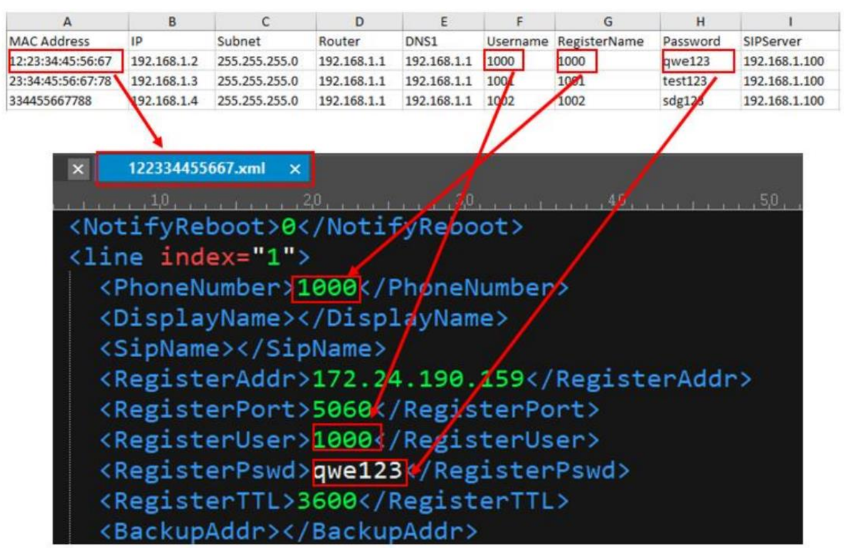
The other parameters in the template.xml file of H2P are default parameters, there is no need to edit if not necessary.

Note The parameter can be customized, but the parameter name in the template.xml and input.xlsx must be matched for each other.

5. Double click “run.exe” after completing the processes above. The configuration files were generated when “Please click anything to close me” is displayed in the bottom line, just click any key to exit this program, as the picture shown below:



6. Enter the folder “out”, all of the configuration files generated are here. Open any one of the configuration files, then check the value of the parameters are the same as the one defined in the “input.xlsx” file, as the picture shown below:



28.4 More

For other issues related with the mass generation of configuration files, please refer to the ALE SIP DeskPhones Configuration File Mass Generation and Installation Guide.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

28.5 Supported Models

H2P.

28.6 Firmware Version

All Version

29 Conclusion of common status codes in SIP messages

29.1 Issue Summary

This FAQ shares the conclusion of common status codes in SIP messages which may help customers to locate the issue faster before contacting ALE support team.

29.2 Possible Causes

NA

29.3 How to Resolve

1. 1xx Temporary response:

Temporary response, that is, the response of the message nature, indicates that the server is processing the request, and the final response has not yet been determined. If the server takes more than 200ms to process the request to generate a final response, it should send a 1xx response.

Note 1xx responses are not transmitted reliably. They will not cause the client to send an ACK response. Temporary (1xx) responses can include the message body and the session description.

100 Trying	This response indicates that the request has been received by the next-hop server and that some unspecified action is being taken on behalf of this call (for example, a database is being consulted). This response, like all other provisional responses, stops retransmissions of an INVITE by a UAC. The 100 (Trying) response is different from other provisional responses, in that it is never forwarded upstream by a stateful proxy.
180 Ringing	The UA receiving the INVITE is trying to alert the user. This response MAY be used to initiate local ringback.
181 Call is Being Forwarded	A server MAY use this status code to indicate that the call is being forwarded to a different set of destinations.
182 Queued	The called party is temporarily unavailable, but the server has decided to queue the call rather than reject it. When the callee becomes available, it will return the appropriate final status response. The reason phrase MAY give further details about the status of the call, for example, "5 calls queued; expected waiting time is 15 minutes". The server MAY issue several 182 (Queued) responses to update the caller about the status of the queued call.
183 Session Progress	The 183 (Session Progress) response is used to convey information about the progress of the call that is not otherwise classified.

	The Reason-Phrase, header fields, or message body MAY be used to convey more details about the call progress.

2. 2xx Successful

The request was successful.

200 OK	The request has succeeded. The information returned with the response depends on the method used in the request.

3. 3xx Redirection

3xx responses give information about the user's new location, or about alternative services that might be able to satisfy the call.

300 Multiple Choices	The address in the request resolved to several choices, each with its own specific location, and the user (or UA) can select a preferred communication end point and redirect its request to that location. The response MAY includes a message body containing a list of resource characteristics and location(s) from which the user or UA can choose the one most appropriate, if allowed by the Accept request header field. However, no MIME types have been defined for this message body. The choices SHOULD also be listed as Contact fields. Unlike HTTP, the SIP response MAY contain several Contact fields or a list of addresses in a Contact field. UAs MAY use the Contact header field value for automatic redirection or MAY ask the user to confirm a choice. However, this specification does not define any standard for such automatic selection. This status response is appropriate if the callee can be reached at several different locations and the server cannot or prefers not to proxy the request.
301 Moved Permanently	The user can no longer be found at the address in the Request-URI, and the requesting client SHOULD retry at the new address given by the Contact header field. The requestor SHOULD update any local directories, address books, and user location caches with this new value and redirect future requests to the address(es) listed.
302 Moved Temporarily	The requesting client SHOULD retry the request at the new address(es) given by the Contact header field (Section 20.10). The Request-URI of the new request uses the value of the Contact header field in the response. The duration of the validity of the Contact URI can be indicated through an Expires (Section 20.19) header field or an expires parameter in the Contact header field. Both proxies and UAs MAY cache this URI for the duration of the expiration time.

	If there is no explicit expiration time, the address is only valid once for recursing and MUST NOT be cached for future transactions. If the URI cached from the Contact header field fails, the Request-URI from the redirected request MAY be tried again a single time. The temporary URI may have become out-of-date sooner than the expiration time, and a new temporary URI may be available.
305 Use Proxy	The requested resource MUST be accessed through the proxy given by the Contact field. The Contact field gives the URI of the proxy. The recipient is expected to repeat this single request via the proxy. 305 (Use Proxy) responses MUST only be generated by UASs.
380 Alternative Service	The call was not successful, but alternative services are possible. The alternative services are described in the message body of the response. Formats for such bodies are not defined here and may be the subject of future standardization.

4. 4XX Request Failure

4xx responses are definite failure responses from a particular server. The client SHOULD NOT retry the same request without modification (for example, adding appropriate authorization).

However, the same request to a different server might be successful.

400 Bad Request	An error occurred in the request, such as the request header is not equal. “Reason Phrase” header should flag this detailed syntax error, such as "Missing Call ID header field"
401 Not authorized	No authentication information is provided. Did not bring token etc. when requesting. This is usually from SIP server while 407 error usually from the outbound proxy server.
403 Access is forbidden	The requested resource is not allowed to be accessed. It means that there is no permission. Usually caused by incorrect username and password.
404 Not found	The configuration file of the phone cannot be found on the server. It may be that the mac address of the phone is incorrect or there is no configuration file of this phone. When the domain of the “Request URI” does not match the domain received, this response will also be generated.
406 Not Acceptable	The content in the request contains an error that cannot be received.
407 Proxy Authentication Required	Similar to 401 (Unauthorized), this error code indicating that the client should first authenticate on the proxy.

408 Request Timeout	For a period of time, the server cannot generate a final response, such as when initiating a registration or call and not receiving a response from the server for a long time.
480 Temporarily Unavailable	The request has successfully reached the called party's terminal system, but the called party is currently unavailable. "Reason Phrase" should provide more detailed reasons why the called party is temporarily unavailable. This value should be able to be set by UA, for example, status code 486 (Busy Here) can be used to more accurately indicate the specific reason for the failure of this request. Many times, it is an error prompt issued when the called end does not support the requesting party's voice media
486 Busy Here	When the called party's terminal system is successfully contacted, but the called party is currently unable to answer the call on this terminal system, such as when the line is busy, DND.
487 Request Terminated	After the request is terminated by BYE or CANCEL, a 487 message will be sent to inform the other end.
488 Not Acceptable Here	This response has the same meaning as 406 (Not Acceptable), but it applies to the specific resource indicated by the Request URI that cannot be accepted, while the request may be accepted elsewhere.

5. 5xx Server Failure

5xx responses are failure responses given when a server itself has erred.

500 Server Internal Error	The server encountered an unexpected condition that prevented it from fulfilling the request. The client MAY display the specific error condition and MAY retry the request after several seconds. If the condition is temporary, the server MAY indicate when the client may retry the request using the Retry-After header field.
501 Not Implemented	The server does not support the functionality required to fulfill the request. This is the appropriate response when a UAS does not recognize the request method and is not capable of supporting it for any user. (Proxies forward all requests regardless of method.) Note that a 405 (Method Not Allowed) is sent when the server recognizes the request method, but that method is not allowed or supported.
502 Bad Gateway	The server, while acting as a gateway or proxy, received an invalid response from the downstream server it accessed in attempting to fulfill the request.
503 Service Unavailable	The server is temporarily unable to process the request due to temporary overloading or maintenance of the server. The server MAY indicate when the client should retry the request in a Retry-After header field. If no Retry-After is given, the client MUST act as if it had received a 500 (Server Internal Error) response. A client (proxy or UAC) receiving a 503 (Service Unavailable) SHOULD attempt to forward the request to an alternate server.

	It SHOULD NOT forward any other requests to that server for the duration specified in the Retry-After header field, if present. Servers MAY refuse the connection or drop the request instead of responding with 503 (Service Unavailable).
504 Server Time-out	The server did not receive a timely response from an external server it accessed in attempting to process the request. 408 (RequestTimeout) should be used instead if there was no response within the period specified in the Expires header field from the upstream server.
505 Version Not Supported	The server does not support, or refuses to support, the SIP protocol version that was used in the request. The server is indicating that it is unable or unwilling to complete the request using the same major version as the client, other than with this error message.
513 Message Too Large	The server was unable to process the request since the message length exceeded its capabilities.

6. 6xx Global Failures

6xx responses indicate that a server has definitive information about a particular user, not just the particular instance indicated in the Request-URI.

600 Busy Everywhere	The callee's end system was contacted successfully but the callee is busy and does not wish to take the call at this time. The response MAY indicate a better time to call in the Retry-After header field. If the callee does not wish to reveal the reason for declining the call, the callee uses status code 603 (Decline) instead. This status response is returned only if the client knows that no other end point (such as a voice mail system) will answer the request. Otherwise, 486 (Busy Here) should be returned.
603 Decline	The callee's machine was successfully contacted but the user explicitly does not wish to or cannot participate. The response MAY indicate a better time to call in the Retry-After header field. This status response is returned only if the client knows that no other end point will answer the request.
604 Does Not Exist Anywhere	The server has authoritative information that the user indicated in the Request-URI does not exist anywhere.
606 Not Acceptable	The user's agent was contacted successfully but some aspects of the session description such as the requested media, bandwidth, or addressing style were not acceptable. A 606 (Not Acceptable) response means that the user wishes to communicate but cannot adequately support the session described. The 606 (Not Acceptable) response MAY contain a list of reasons in a Warning header field describing why the session described cannot be supported.

	A message body containing a description of media capabilities MAY be present in the response, which is formatted according to the Accept header field in the INVITE (or application/sdp if not present), the same as a message body in a 200 (OK) response to an OPTIONS request. It is hoped that negotiation will not frequently be needed, and when a new user is invited to join an already existing conference, negotiation may not be possible. It is up to the invitation initiator to decide whether to act on a 606 (Not Acceptable) response. This status response is returned only if the client knows that no other end point will answer the request.

29.4 More

Reference documentation: “SIP: Session Initiation Protocol” https://datatracker.ietf.org/doc/html/rfc3261

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

29.5 Supported Models

H2P/H3G/H6/M3/M5/M7/M8.

29.6 Firmware Version

All Version

30 How to enter the post mode of H2P

30.1 Issue Summary

This FAQ shares the detailed steps about how to enter the post mode of H2P.

30.2 Possible Causes

The H2P is stuck at the power up page.

30.3 How to Resolve

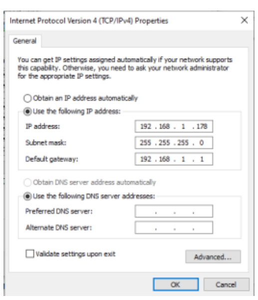
1. Power on the phone, wait for the power LED blinking red;

Note If the LED never blinks red, then this FAQ can't be used, contact ALE support team for further support

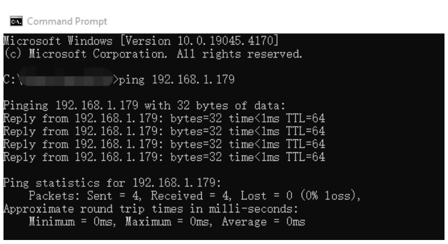
2. Press the "#" key;
3. Phone will enter the Post Mode, showing an IP address;

4. Set static IP on your PC with the same subnet of the phone;

Eg:



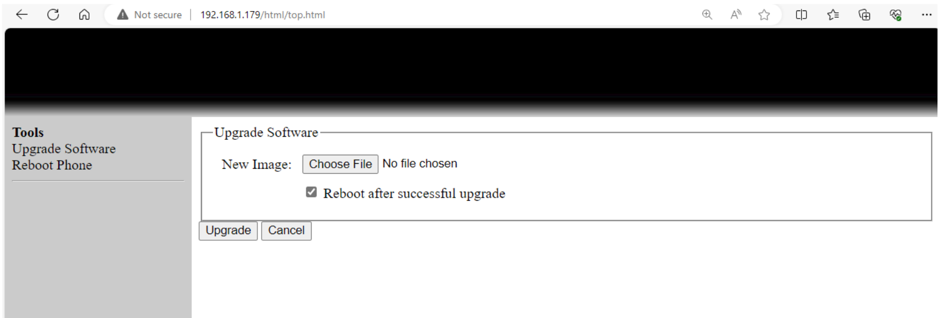
5. Connect the PC to the LAN port of the phone;
6. Ping the phone IP to make sure it is working;



7. Open browser of PC, enter "http://192.168.1.179" to enter the phone web page;

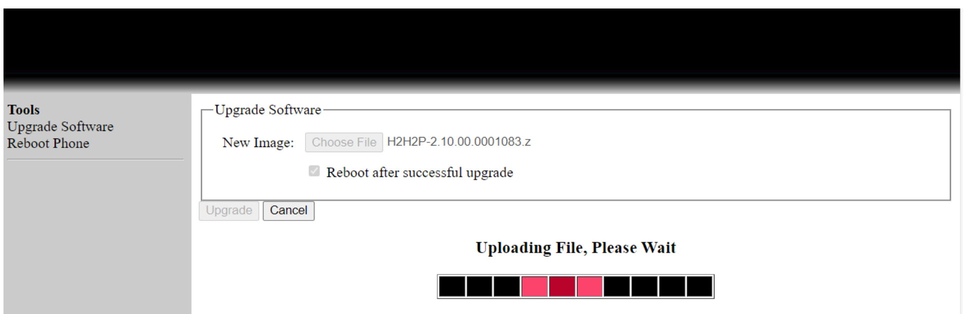
8. Go to “Upgrade Software”;

9. Click “Choose File” to select a binary file of the phone;



10. Click “Upgrade” to start the upgrade process for H2P DeskPhone;

Note If any message shows like “Filename does not match”, just ignore it and click “OK” to continue



11. After the upgrade is finished, it should be OK to power up.

30.4 More

You can refer to this demonstration video to get more information on the operation: [How to enter H2P post mode].

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

30.5 Supported Models

H2P.

30.6 Firmware Version

All Version

31 How to deploy ALE Myriad series DeskPhone to Teams Gateway

31.1 Issue Summary

This FAQ shares the detailed steps about how to deploy ALE Myriad series DeskPhone to Teams Gateway to make the phone registered to use the Teams Gateway features.

31.2 Possible Causes

Quick deployment guide for ALE SIP DeskPhones with Teams Gateway.

31.3 How to Resolve

1. Configure SIP Gateway following with below link in order to get the auto provision URL for the phones: https://learn.microsoft.com/en-us/microsoftteams/sip-gateway-configure
2. Click here to download the auto provisioning guide for the Myriad series DeskPhones
3. After provisioned, a “Sign-in” softkey will appear at the left side of the softkeys under the LCD screen.
4. Press Sign-in on the SIP DeskPhone to display the authentication URL and pairing code. The pairing code is time sensitive. If it expires, the user must press Back on the phone and start the sign-in process again.
5. Navigate to the authentication URL on the user's desktop or mobile browser and use corporate credentials to log in.
6. Enter the pairing code displayed on the SIP DeskPhone into the web authentication app to pair the SIP phone with the user's account. On a successful sign-in, which might take a while, the SIP phone will display the phone number and username which means the phone is registered and connected to Teams Gateway successfully!

31.4 More

For the features supported by Teams Gateway, please click here to view the “Benefits of SIP Gateway” part; For other issues related with Teams SIP Gateway, please refer to the Troubleshooting Guide for ALE Myriad DeskPhones with Teams SIP Gateway.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

31.5 Supported Models

M3/M5/M7/M8.

31.6 Firmware Version

M3/M5/M7: 2.14.03.000.2345

M8: 2.14.05.000.2352

32 H2P can't receive IP calls but can call out IP calls

32.1 Issue Summary

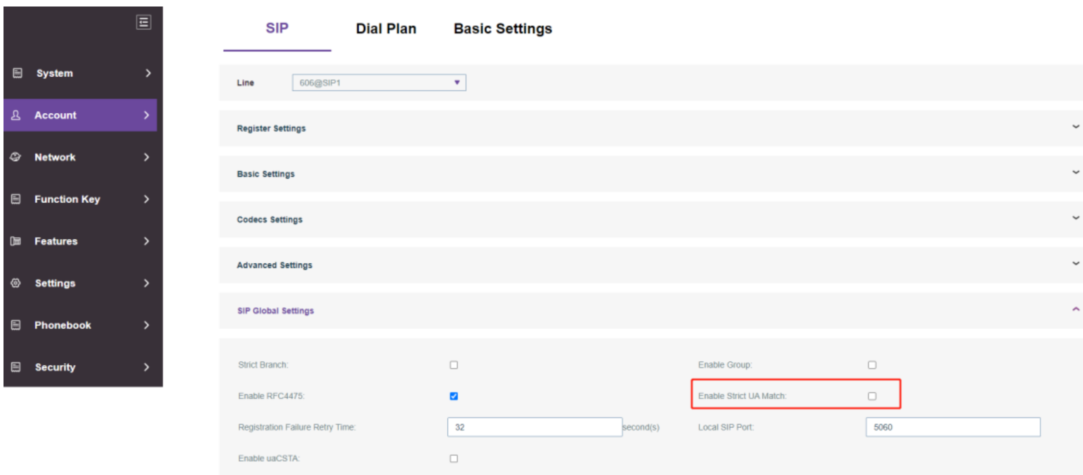
Customers usually meet the problem that H2P can call out IP calls but can’t receive IP calls.

32.2 Possible Causes

Configuration issue.

32.3 How to Resolve

1. Login to phone web UI of H2P
2. Go to Account -> SIP Global Settings, find “Enable Strict UA Match” and then disable it.



3. Test again and the H2P can receive the IP call now.

32.4 More

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

32.5 Supported Models

H2P.

32.6 Firmware Version

H2P: 2.10.000.0001083 and above

33 How to recover H3G/H6/MX Deskphones by firmware rollback

33.1 Issue Summary

When the phone stays at the booting interface all the time and cannot enter the operation interface, this FAQ will help you to recover the phone by firmware rollback.

33.2 Possible Causes

Some program of phone crash suddenly, or the phone gets stuck during the upgrading as some incorrect operations.

33.3 How to Resolve

1. Long press the “1”, “3”, “8”, “0” keys at the same time, then power up the phone.
2. Release the above keys when all the line key LEDs at the side of the screen are blinking.
3. Wait for the firmware to rollback and reboot automatically.

33.4 More

You can refer to this demonstration video to get more information on the operation: How to recover H3G, H6 & MX by firmware rollback.

If still any issue you meet, please feel free to contact ALE support team for further support at support.alesip@al-enterprise.com.

33.5 Supported Models

H3G/H6/M3/M5/M7/M8.

33.6 Firmware Version

All version

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 Technical Documentation Library.
- You have read through the related troubleshooting guides and technical bulletins available in the Technical Documentation Library.
- You have read through the self-service information on commonly asked support questions and known issues and workarounds available in the Technical Knowledge Center.

- END OF DOCUMENT -