Administration Guide

Document Version: 1.1
Software Version: 1.3.xxx.xxx
Release Date: 2016/01/31

Table of Contents

CHAPTER 1.OVERVIEW

The target audiences of this document are IT administrators, technical professionals, and technical supporters. This document assumes the audiences have preliminary knowledge about Internet, network infrastructures and protocols, Voice over IP technologies, etc. Hence, this document will not explain particular protocols or standards in detail.

The administration guide is to help IT administrators to manage massive phones in an enterprise or small office environment. An IT administrator may follow the guidance in this document to set up a provisioning server to massively configure tens, hundreds, or thousands of phones through the network with pre-scripted configuration files. Meanwhile, the IT administrators can learn how to enhance the quality of service (QoS) , or set up a local upgrade server, etc., from this document.

CHAPTER 2. AUTO-PROVISIONING – FOR MASSIVE DEPLOYMENT OF PHONES

2.1 Introduction

This section provides instructions on how IP phones interoperate with provisioning server for auto provisioning, and shows you four major tasks to provision the phones. It will help users who are not familiar with auto provisioning to understand this process more easily and quickly.

There are 4 configuration files both of which are CFG formatted that the phone will try to download from the server during provisioning.

a. Common CFG file
b. MAC-oriented CFG file
c. Custom-named CFG file
d. ID-oriented CFG file

The Common CFG file will be effectual for all the phones of the right model. A common CFG file has a fixed name for each model，The names of the Common CFG file for each model are:

VIP-1120PT: f0VIP-1120PThw1.100.cfg
VIP-2140PT： f0VIP-2140PThw1.100.cfg

The common config file is very helpful for taking Auto Provision deployment to mass terminals. For example, if you would like to update firmware for 1000 terminals of VIP-2140PT automatically, you will just need a common config file of f0VIP-2140PThw1.100.cfg) containing firmware parameters with deployment, then put it on the appropriate server which the configured Auto-Provison process used.

The device config file named after MAC address is just effective for the terminal with the corresponding MAC address. Its name is the MAC address to remove connectors. For example, the MAC address of VIP-2140PT is "00: 30: 4f: 11: 3a: f8", so the config file name is "00304f113af8.cfg".

The custom-named config file means that users are able to customize its name, For example, user names a device's config file name as "name.cfg", then the phone will go to request and download the common config file and name.cfg from the relevant server.

Sip PnP and DHCP Option can specify the config file name according to the URL, such as http:// user: password@192.168.2.2/name.cfg or http:// user: password@192.168.2.2/$input.cfg. The second method is to let the user enter a file name via the LCD. If you do not specify http:// user: password@192.168.2.2 or http:// user: password@192.168.2.2 $ mac.cfg , it will be named after Mac address.

2.2 Provision Methods

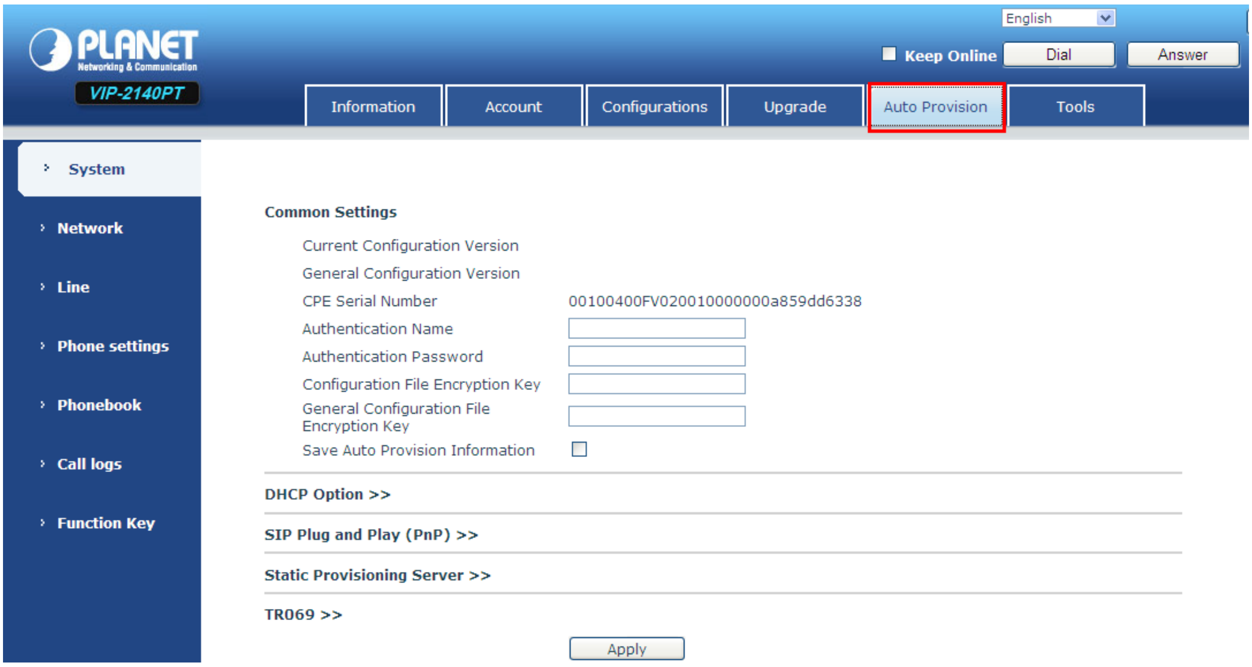
PLANET IP phones support obtaining the provisioning server address in the following ways:

- Plug and Play (PnP) Server
- DHCP Options
- Static Provisioning Server

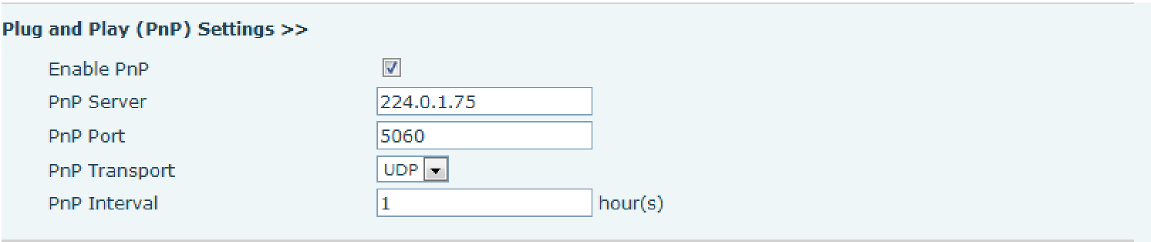
The priority of obtaining the provisioning server address is as follows:

PnP Server --> DHCP Options --> Static Provisioning Server.

You can find the configuration in the Web / System / Auto provision.



2.2.1 Plug and Play (PnP) Server



As shown:

Enable PnP: whether to start SIP PnP

PnP Server: pnp server address

PnP Port: pnp server port number

PnP Transport: Transport Protocol

PnP Interval: time interval polling to check the time interval between two requests

Restart the phone after setting. It will send the subscribe package to the pnp server, then the server returns the notify package with URL. The phone parses URL after receiving it.

If you start the PnP server for push deployment, it will periodically send SUBSCRIBE message to the server after the terminal starts. Take VIP-2140PT for example, the format of Message Header of SIP SUBSCRIBE message is as follows.

Via: SIP/2.0/UDP 192.168.1.45:5060;branch=z9hG4bK3102710241234624733

From: <sip:MAC=00304fa99948@224.0.1.75>

To: <sip:MAC=00304fa99948@224.0.1.75>

Call-ID: 322432620212850-163241588724467@192.168.1.45

CSeq: 1 SUBSCRIBE

Contact: <sip:192.168.1.45:5060>

Max-Forwards: 70

User-Agent: PLANET VIP-2140PT

Expires: 0

Event:ua-profile;profile-type="device";vendor="PLANET";model="VIP-2140PT";version="2.0.3.2991"

Accept: application/url

Content-Length: 0

Any SIP servers compatible with the particular message server will respond and send back a SIP NOTIFY message including the server URL of Auto Provision. The Message Header of SIP NOTIFY message is as follows:

Via: SIP/2.0 / [transport] [local_ip]: [local_port]; branch = [branch]

From: <sip:MAC= 00304fa9994a192.168.1.169>

To: <sip:MAC= 00304fa9994a192.168.1.169>

Call-ID: 176851610432700-321342882818040@192.168.1.14

CSeq: 3 NOTIFY

Max-Forwards: 70

Content-Type: application / url

Subscription-State: terminated; reason = timeout

Event: ua-profile; profile-type = "device"; vendor = "PLANET"; model = "VIP-2140PT"; version = "2.0.3.2991"

Content-Length: 29

http://192.168.1.118/ $ mac.cfg

http://192.168.1.118/$mac.cfg in the NOTIFY message is the URL of the config file to be downloaded.

Note:

PnP mechanism can support two forms of $mac and $input to get the config file, as well as server username and password authentication.

2.2.2 DHCP Options

WAN Mode of the phone must be DHCP to use DHCP Option.

There are four options available for DHCP Option: DHCP option 66, DHCP option 43, Custom DHCP Option and DHCP Option Disable.

Custom DHCP Option's setting range is 128~254. DHCP Option Disable means closing DHCP Option.

Restart the phone or wait for a renewal of DHCP server after setting, then the phone will request for option information from the DHCP server. If the server replies the option information you requested, you will see the corresponding option information from the capture package replied from the server. Filter "bootp" and view the ACK package to get the URL which the phone will parse.

When obtaining an application parameter of Auto Provision through DHCP Option mode, the user can optionally choose one option. For example, If DHCP option 43 is selected, it will have the following field values in the "DHCP discover message" and "DHCP request message" which terminal sends to the server.

Option: (t=55,l=7) Parameter Request List

Option: (55) Parameter Request List

Length: 7

Value: 011c0302042b06

1 = Subnet Mask

28 = Broadcast Address

43 = Vendor-Specific Information

It will have the following field values in the "DHCP offer message" and "DHCP ACK message" which the server sends to the terminal.

Option: (t=43,l=29) Vendor-Specific Information

Option: (43) Vendor-Specific Information

Length: 29

Value: 746674703a2f2f3139322e3136382e312e3131382f246d61...

"Value" in "Option: (t=43, l=29) Vendor-Specific Information" is the hexadecimal form of the URL which is the path to download the config file. The Value is http://192.168.1.118/$mac.cfg. PLANET terminal supports $mac replacement. The value of the URL can be http://ip/$mac.cfg or http://ip/mac.cfg?mac=$mac.cfg

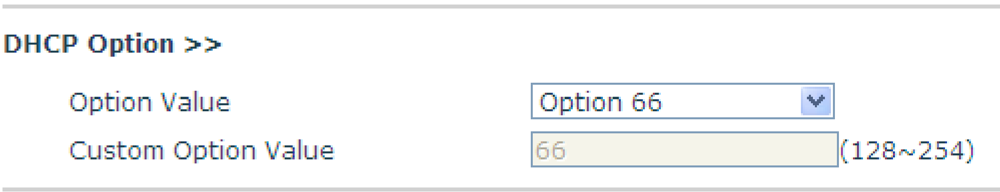
"DHCP option 66" and "DHCP custom option" application parameters are similar to the DHCP option 43 above.

Note:

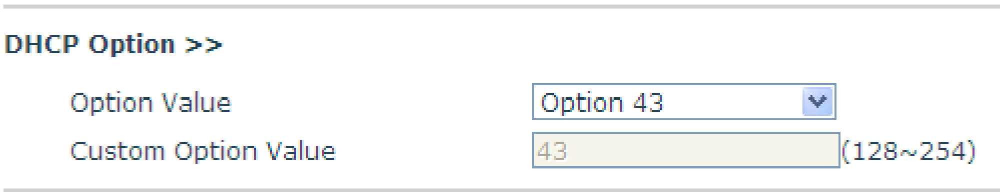
PLANET devices also support URL form of http://ip/$input.cfg. If the "Value" in the above "Option: (t = 43, l = 29) Vendor-Specific Information" is http://192.168.1.118/$input.cfg, the phone will pop up a dialog box of inputting devices' corresponding configuration ID values. The ID value is assigned by the administrator. After entering the devices' corresponding configuration ID values, the devices will automatically download the config file of the corresponding ID from the server. PLANET devices support $input replacement simultaneously. The value of the URL can be http://ip/$input.cfg or http://ip/input.cfg?input=$input.cfg.

Some HTTP/HTTPS/FTP servers require authentication of username and password, then PLANET devices have two methods for this. Firstly, you can add username and password in the URL, for example, http://username:passwd@ip/$mac.cfg. Secondly, for URL without username and password, the devices will pop-up a dialog box to request user to enter username and password for authentication.

Option 66：

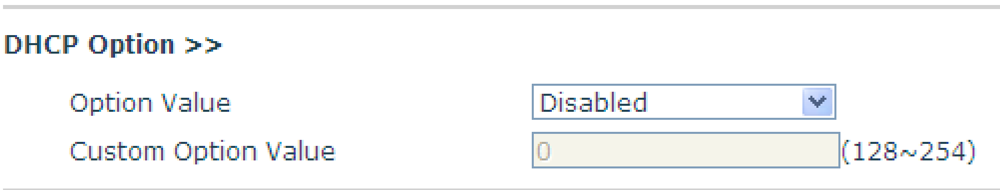


Option 43：



Custom DHCP Option:

DHCP Option Disable：



2.2.3 Static Provisioning Server

As shown:

Server Address: The address of the server can carry the path where the config file is stored. Enter username and password the server requires and the port number of the server. Username and password can be filled in the web page or entered on the LCD when downloading the config file.

Config File Name: The name of the device config file is the same with the Configuration name described in the URL section. You can refer to the URL section for details.

Protocol Type: There are four kinds of servers that can be selected: ftp, tftp, http and https

Update Interval: Time interval between two downloads.

Update Mode: There are three types of Static Provisioning Server update mode: Disable, Update After Reboot and Update at Time Interval.

2.3 Server Setup

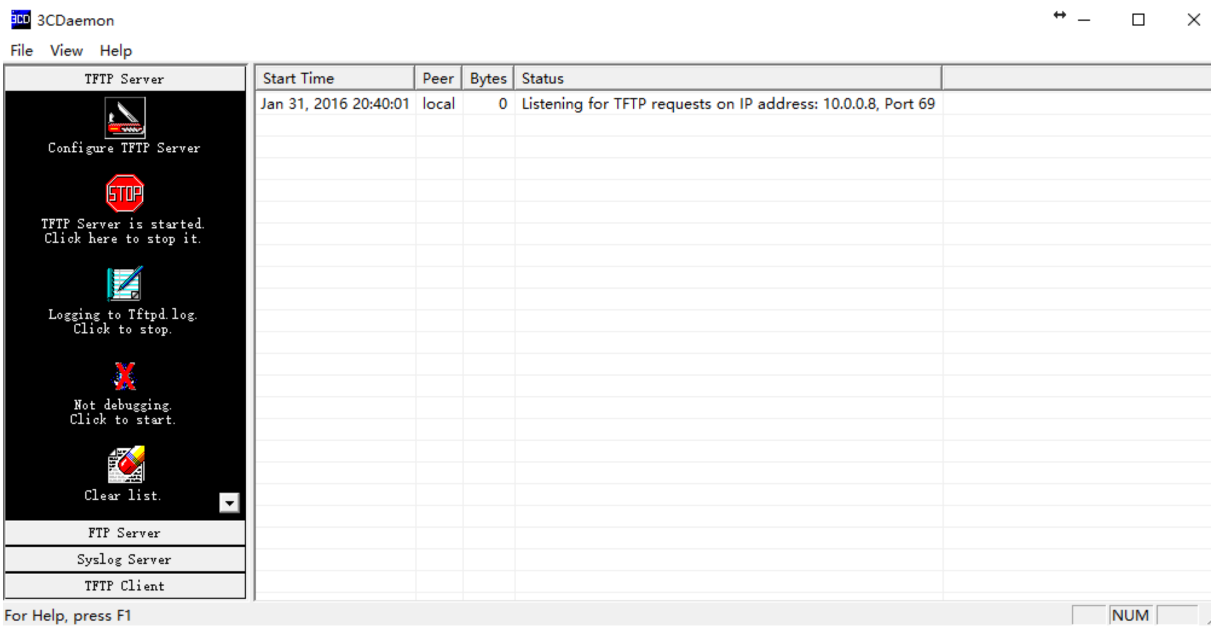
VIP-1120PT and VIP-2140PT products support using FTP, TFTP, HTTP and HTTPS protocols to download configuration files. You can use one of these protocols for provisioning. Users can choose a suitable method to deploy devices.

2.3.1 Configure a TFTP server

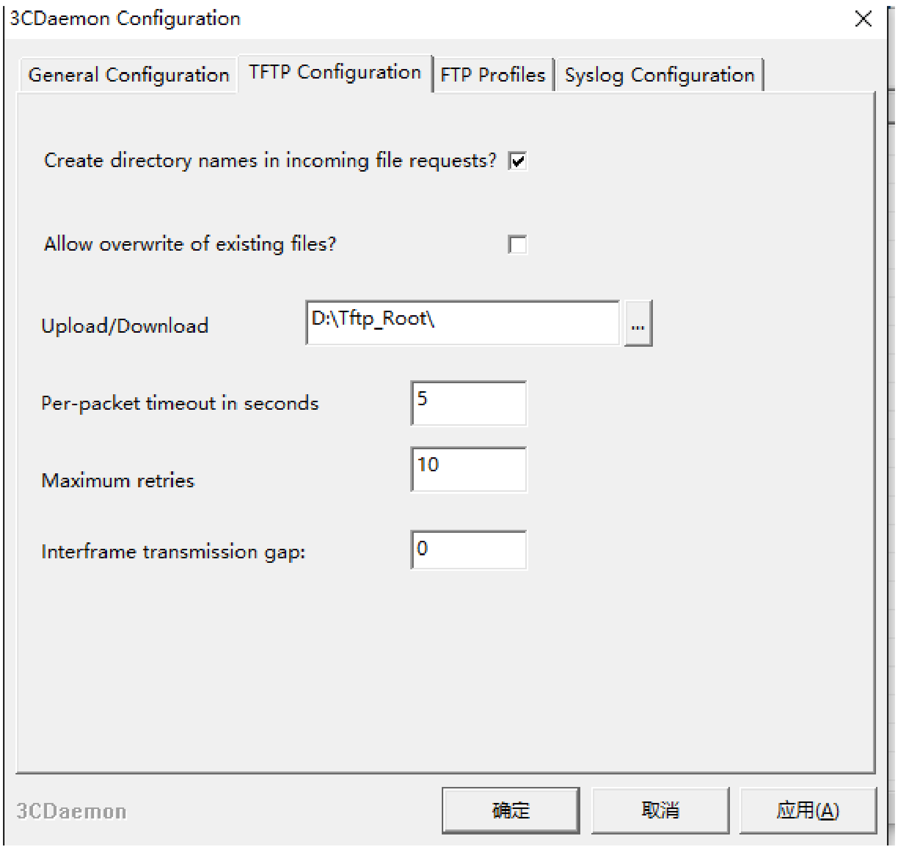
The following section provides instructions on how to configure a TFTP server.

We recommend that you use 3CDaemon or TFTPD32 as a TFTP server. Both of them are free applications for Windows. You can download 3CDaemon online:
http://www.oldversion.com/3Com-Daemon.html and TFTPD32 online: http://tftpd32.jounin.net/.

1. First, install 3CDaemon if it is not already on your application server host.



2. Select **Configure TFTP Server.** Click the button  to locate the TFTP root directory from your local system:



3. Click the **Confirm** button to finish configuring the TFTP server.

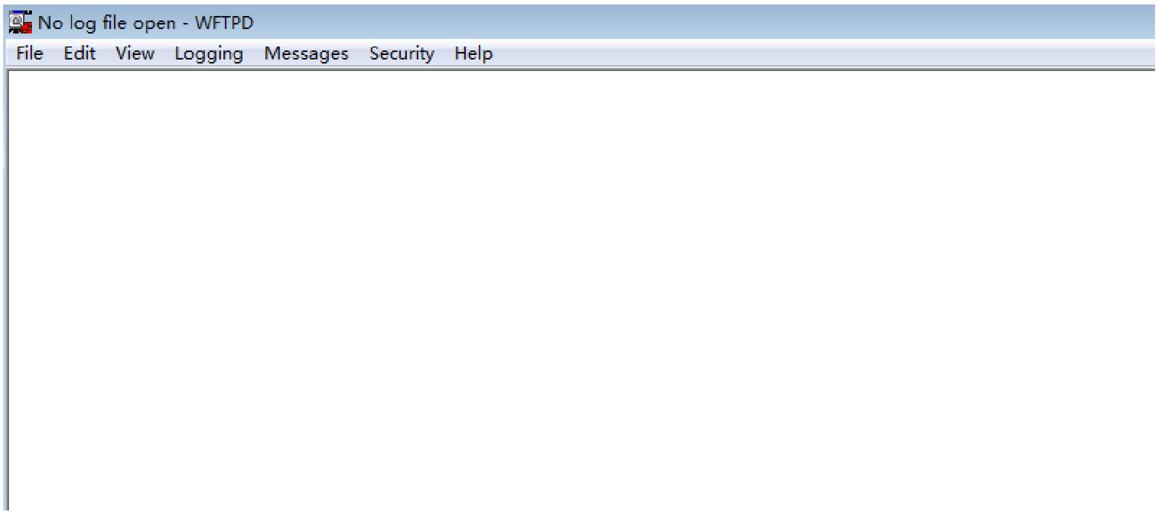
The server URL "tftp://IP/" (Here "IP" means the IP address of the provisioning server, for example, "tftp://10.0.0.3/") is where the IP phone downloads configuration files from.

2.3.2 Configure an FTP server

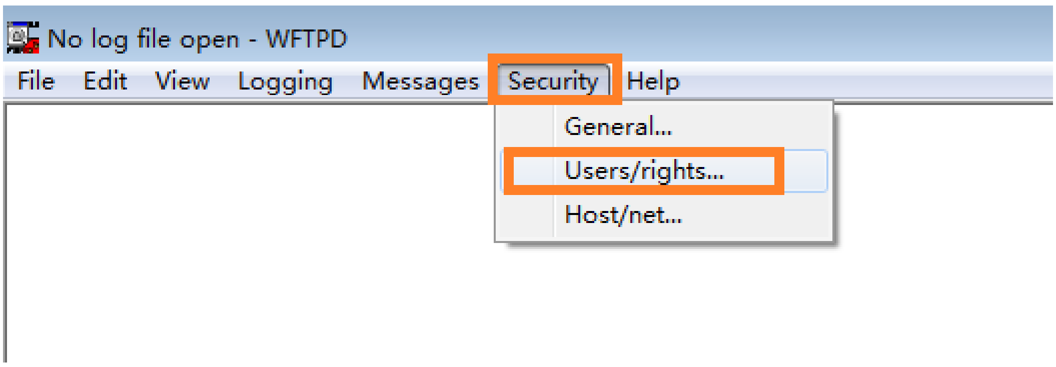
If there is a wftpd application installed on your PC, open it now, or otherwise, download and install it.

To configure an FTP server:

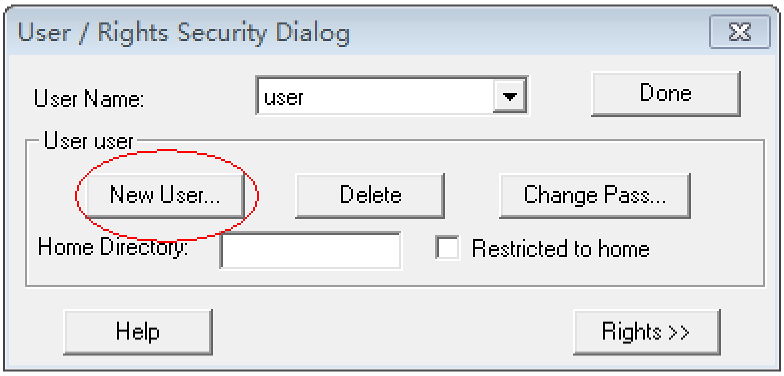
1. Double-click the wftpd.exe to open the application.
2. A screenshot is shown below:



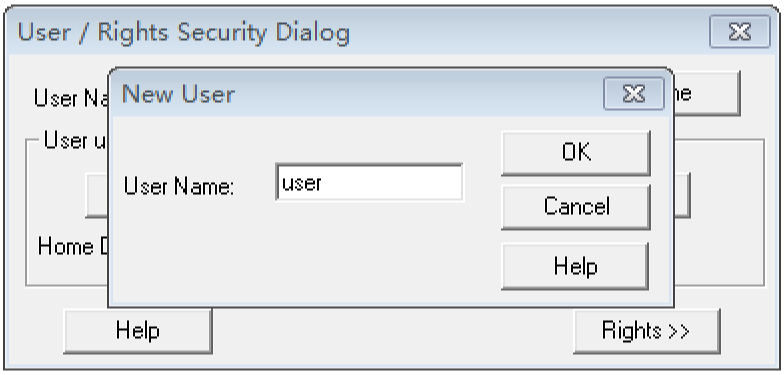
3. Select **Security**, then click **User/right**



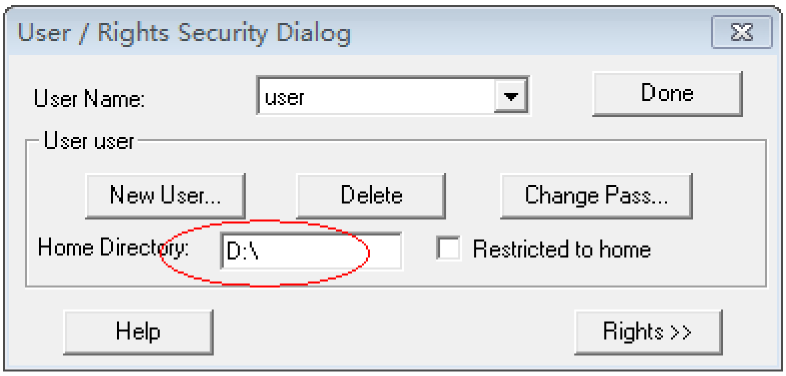
4. Click **New User,**



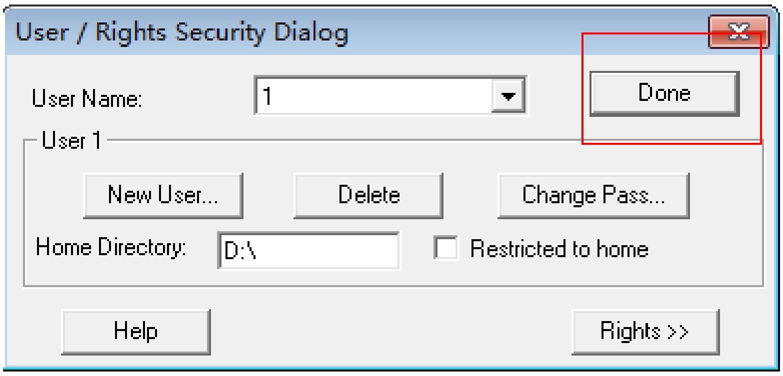
5. Enter the new authentication username in the Profile field.



6. Enter the path to locate the FTP root directory on the computer:



7. Click the **Done** button to save the settings and finish the configurations.



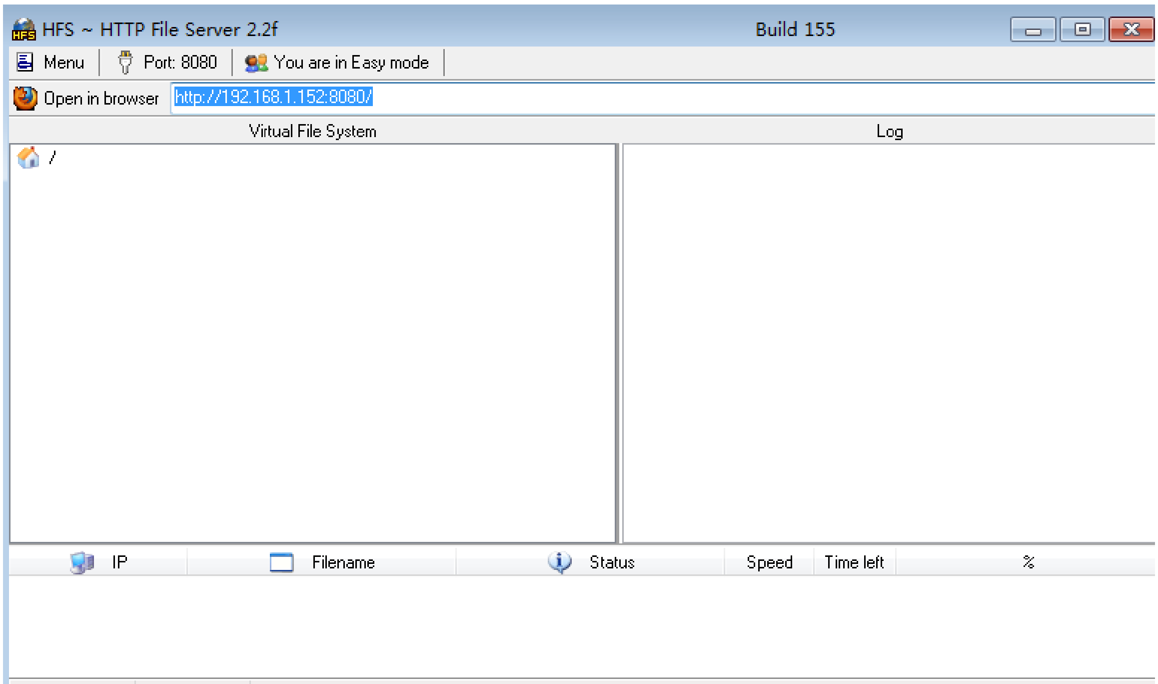
The server URL "ftp://username:password@IP/" (Here "IP" means the IP address of the FTP server, "username" and "password" are the authentication for FTP server access.

For example, ftp://123:123@10.0.0.3/)

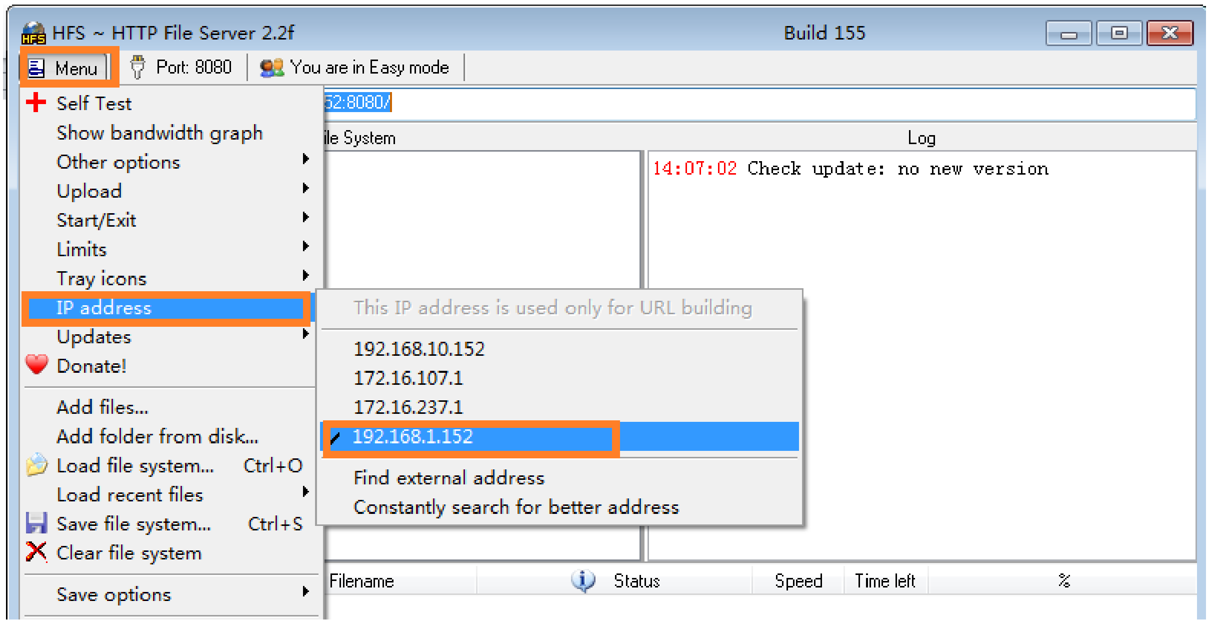
2.3.3 Configure an HTTP server

1. First, install 3CDaemon if it is not already on your application server host.
If there is a hfs application installed on your computer, open it now, or otherwise, download and install it.
To configure an HTTP server:

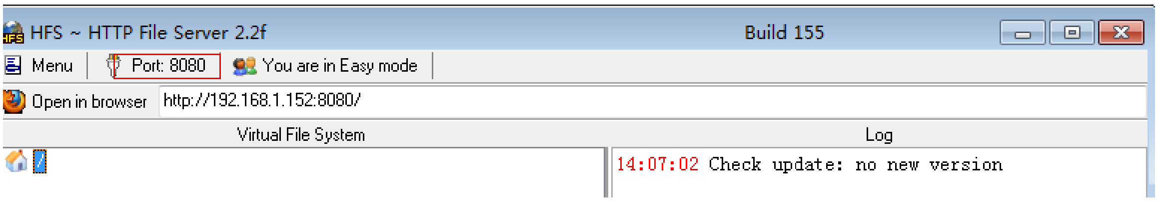
1. Double-click the hfs.exe to start the application.
2. A screenshot is shown below:



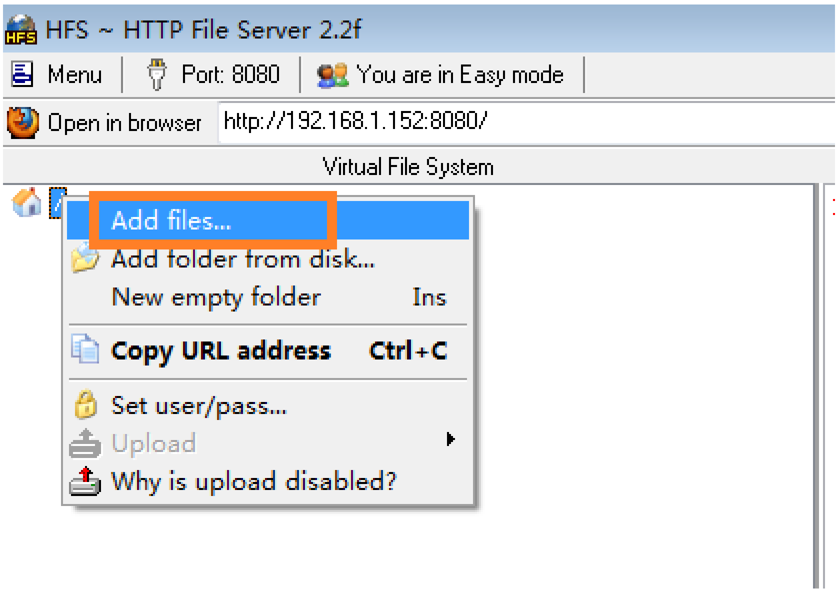
3. Click **menu**, then select **the IP address** to choose the PC’s IP.



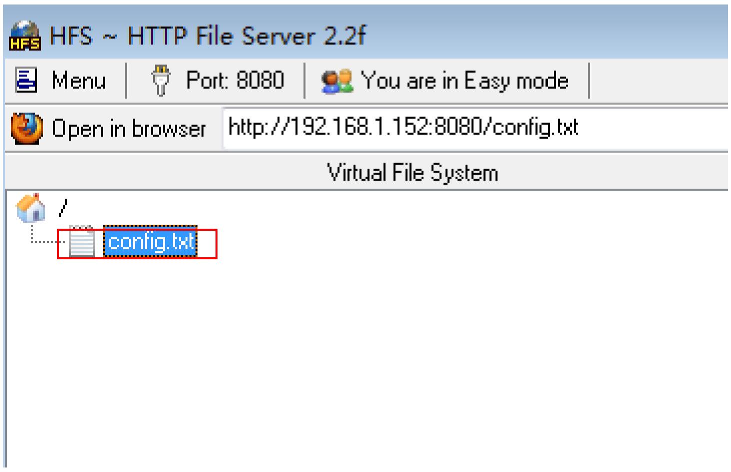
4. The default HTTP port is 8080. You can also reset the HTTP port (make sure the port isn’t used before you reset).



5. Right-click the icon on the left of the main page. Select Add folder from disk to add the HTTP Server root directory.



6. Locate the root directory from the computer system. Select the kind of folder which you deployed.

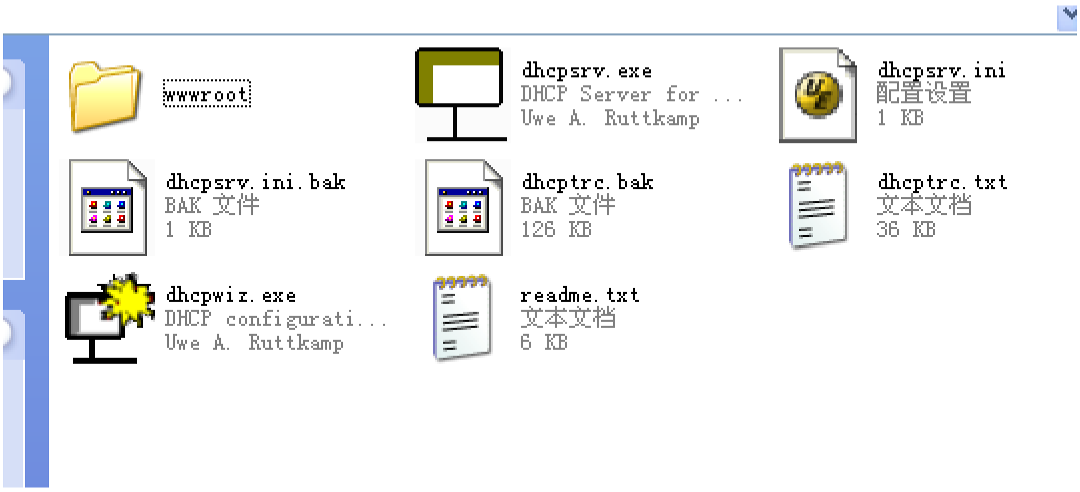


7. Check the server URL “http:// IP:Port/” in the “Open in browser” address bar (For example, the server URL ”http:// 192.168.1.152:8080/” is shown on the screenshot). We recommend that you can fill out the server URL in the address bar of the web browser and then press <Enter> key to check the HTTP server before provisioning.

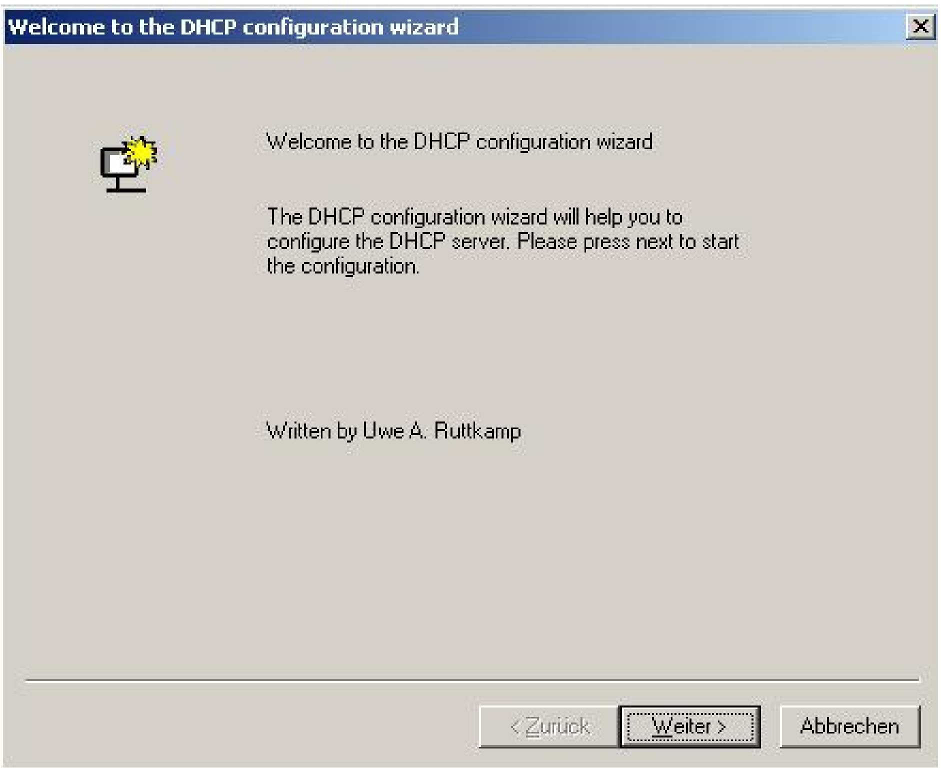
2.3.4 Configure a DHCP server

Before configuring the DHCP Server, please make sure that:
There is no DHCP server in your local system. Or it will cause some unknown consequences on deploy procedure.
If you have a dhcpsrv2.3 application installed on your computer, open it now, or otherwise, download and install it.
To configure a DHCP server:

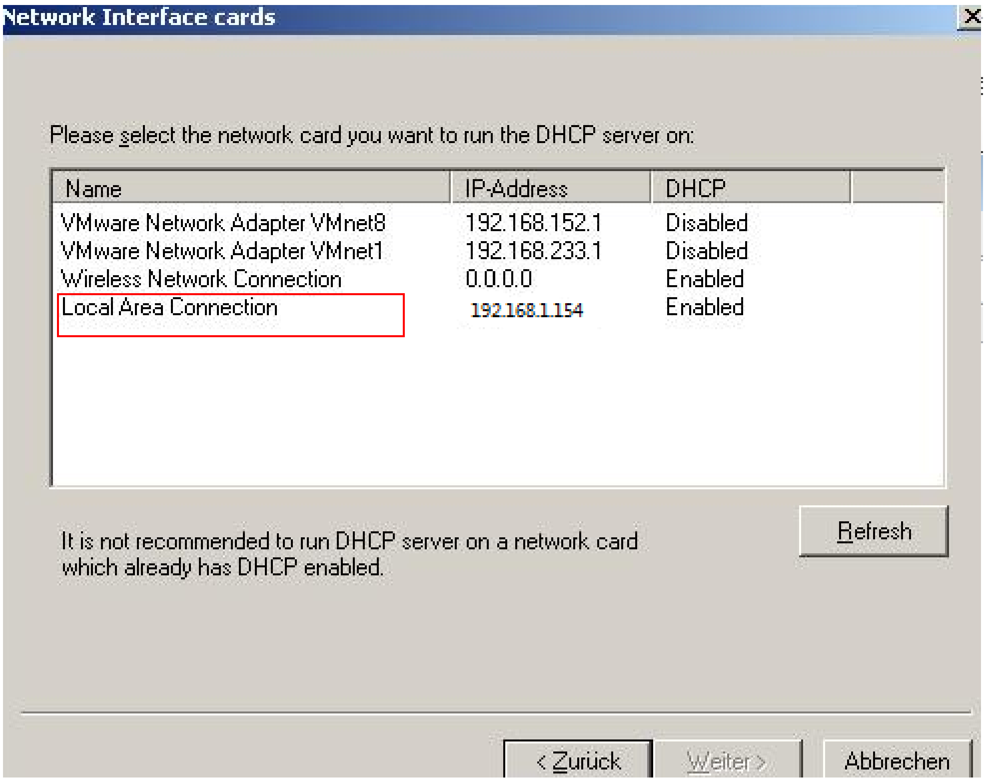
1. Open the dhcpsrv2.3 folder



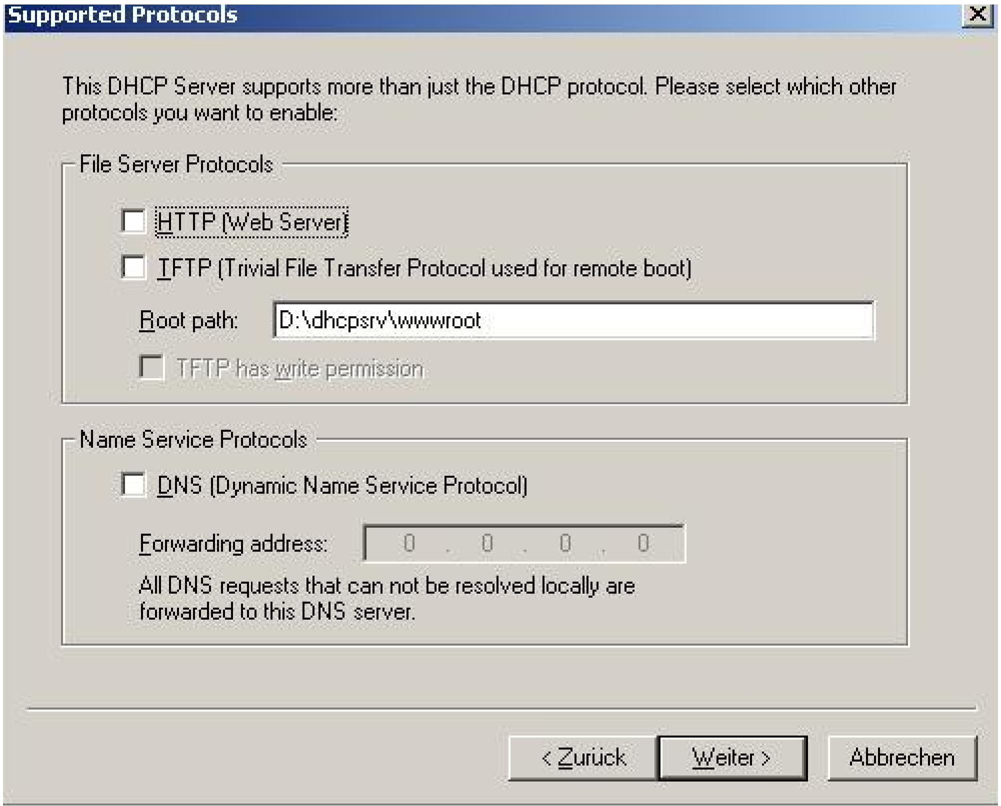
2. Open the application “**dhcpwiz.exe**” to configure the DHCP Server.



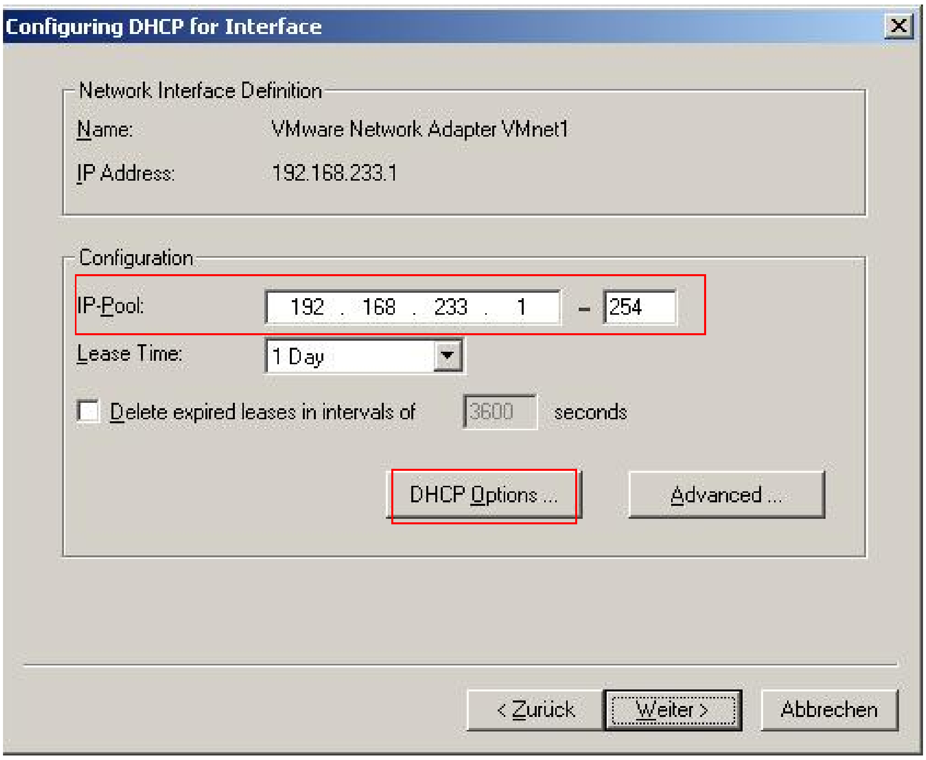
3. Click "**Weiter**", and select "**Local Area Connection**", and then click "**Weiter**".



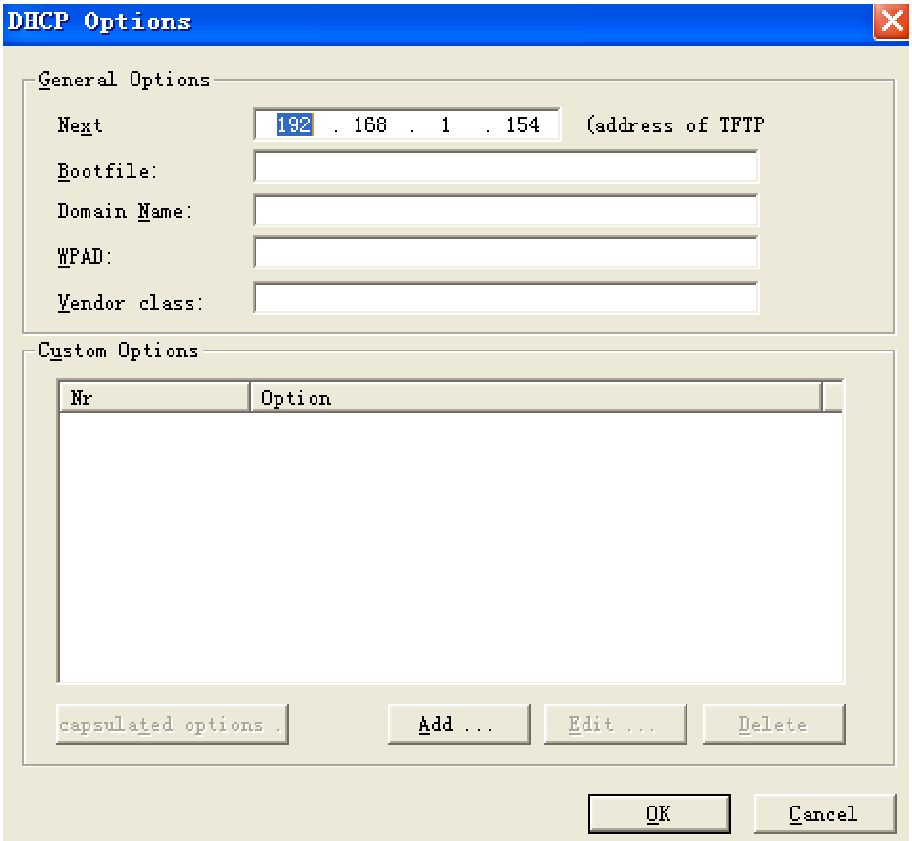
4. Click "**Weiter**"



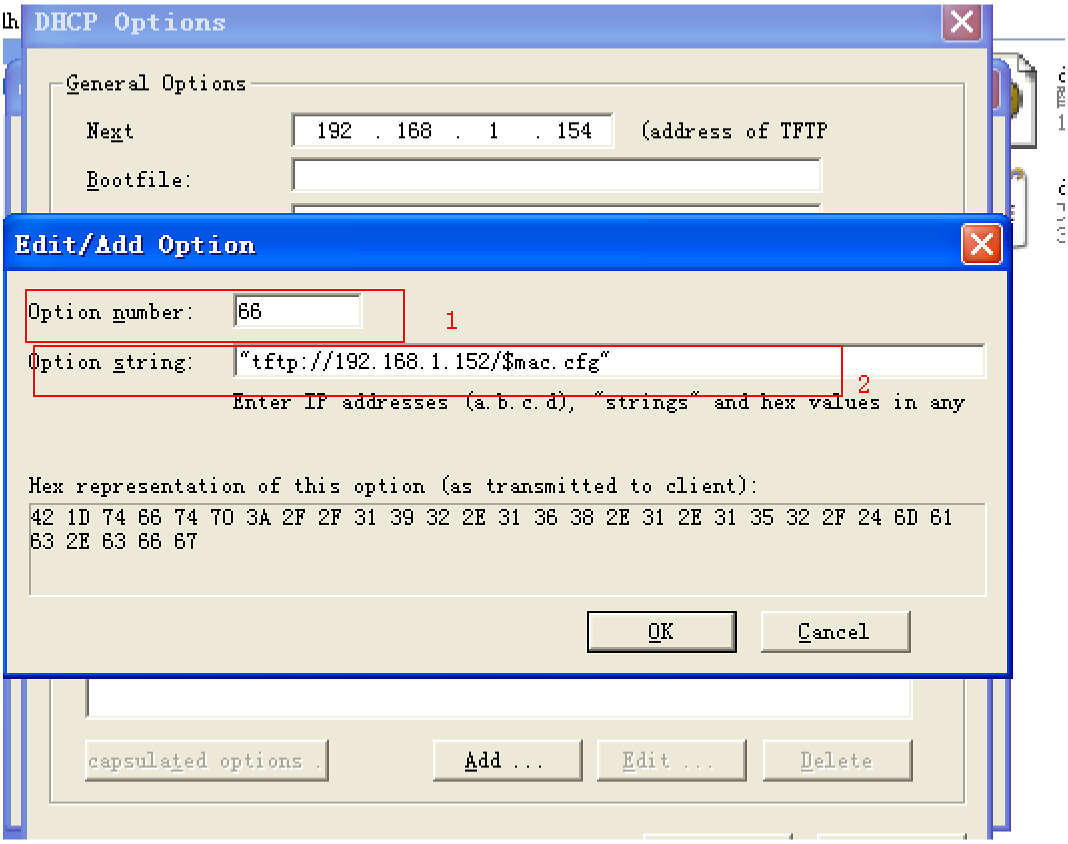
5. Set the DHCP IP range, and click "**DHCP Options**".



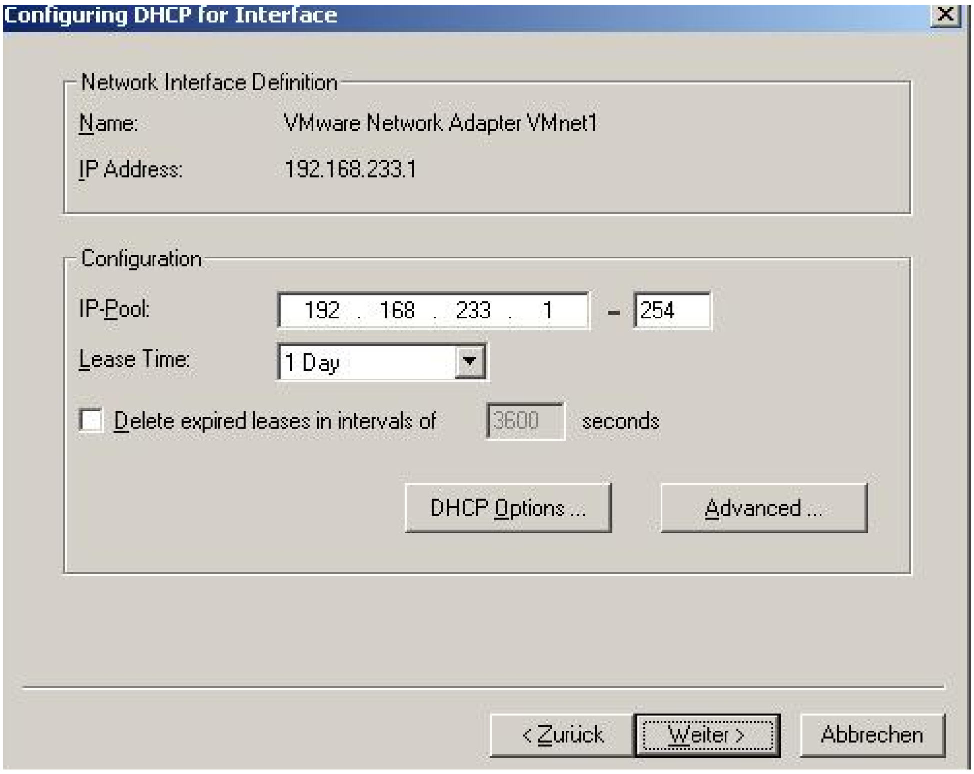
6. Click "**Add**"



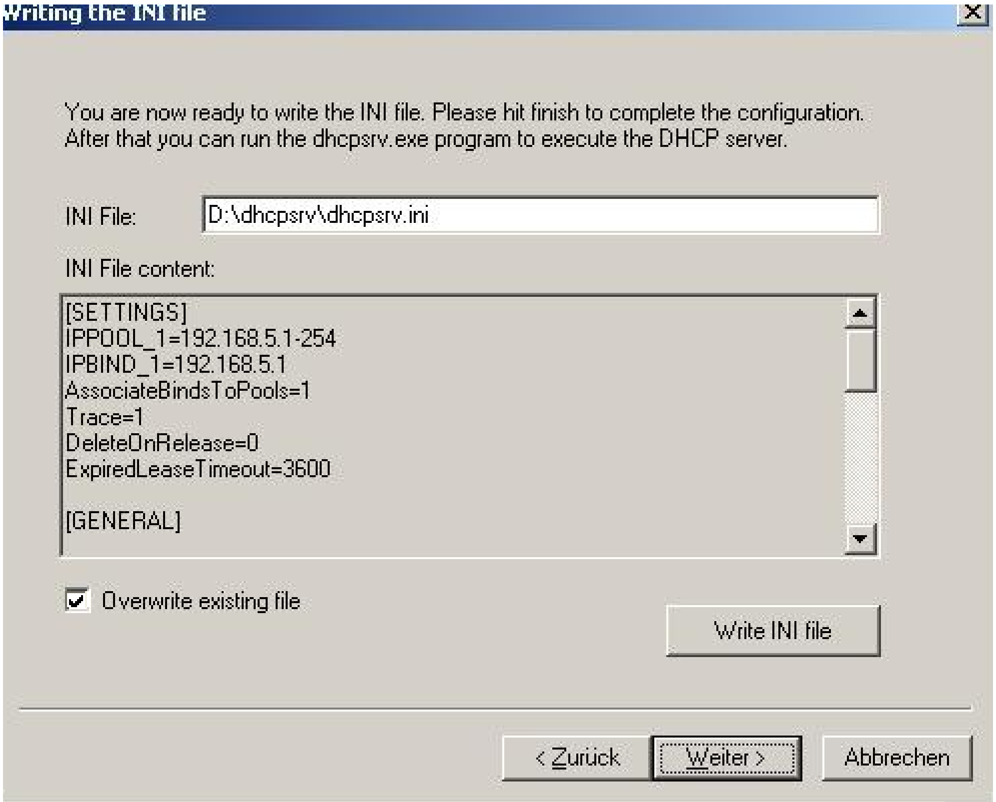
7. You can add the needed DHCP option value, and fill out the provisioning server address in the input field. Take option 66 for example.



8.Click “**OK**”, and then click “**Weiter**”.

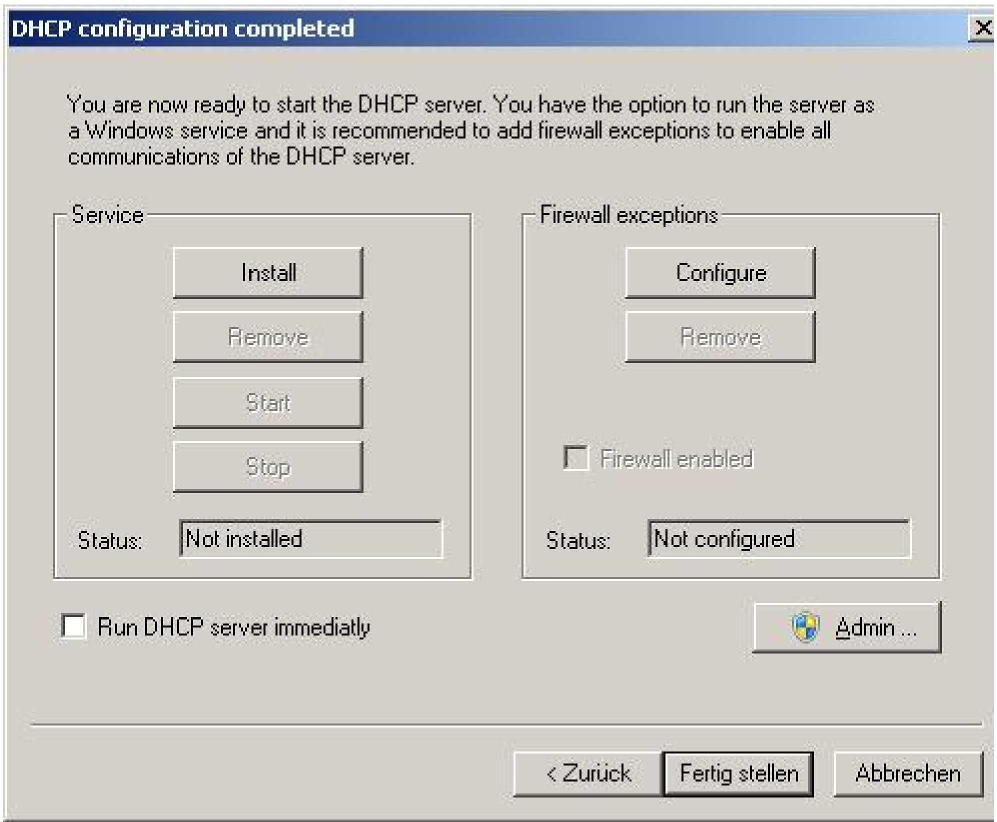


9. Select “**Overwrite existing file**”, and then click “**Write INI file**”.



10. Click “**Install**”, and then click “**start**”.

11. Click “**Fertig stellen**” to accomplish the DHCP option configuration.



Note: Follow the above steps to finish setting the DHCP Server. If you use DHCP Server, just double-click

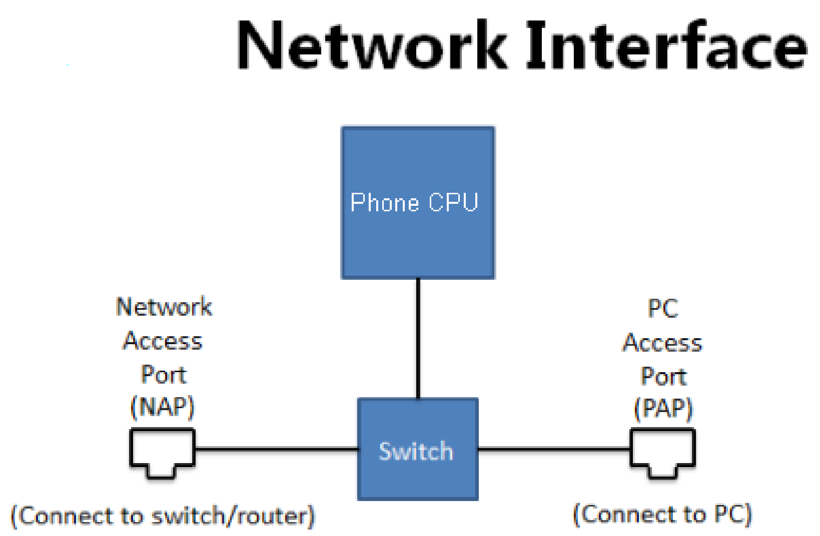
CHAPTER 3. QUALITY OF SERVICE (QOS) – ENHANCE THE COMMUNICATION QUALITY

3.1 VLAN

3.1.1 Introduction

VLAN (Virtual Local Area Network) is used to logically divide a physical network into several broadcast domains. VLAN membership can be configured through software instead of physically relocating devices or connections. Grouping devices with a common set of requirements regardless of their physical location can greatly simplify network design. VLANs can address issues such as scalability, security and network management.

VIP-1120PT/VIP-2140PT consists of two network interfaces – Network Access Port (NAP) and PC Access Port (PAP), and an internal switch works in bridged mode. The NAP should be connected to the LAN network or Internet via a switch or router. The PAP is to provide connected PC to access network or VIP-1120PT/VIP-2140PT.



When VIP-1120PT/VIP-2140PT is configured with an VLAN ID, the VLAN ID will be applied on the outbound packets sent from VIP-1120PT/VIP-2140PT core CPU. This includes all packets, such as ARP, ICMP, DHCP, DNS, HTTP(S), FTP, SIP, etc.

For packets of which destination is not to VIP-1120PT/VIP-2140PT core CPU, it will be copied to the other port and will not be added any VLAN ID, nor to remove any. In another words, packets to and from connected PC will be kept as they are.

While VIP-1120PT/VIP-2140PT tags its outgoing packets with configured VLAN ID, it can accept and process packets with any VLAN ID or no VLAN ID for both NAP and PAP as long as the destination is to VIP-1120PT/VIP-2140PT core CPU address.

3.1.2 Configure VLAN for VoIP Priority

QoS is for mobile phone data packets. It is able to distinguish between voice and data application.

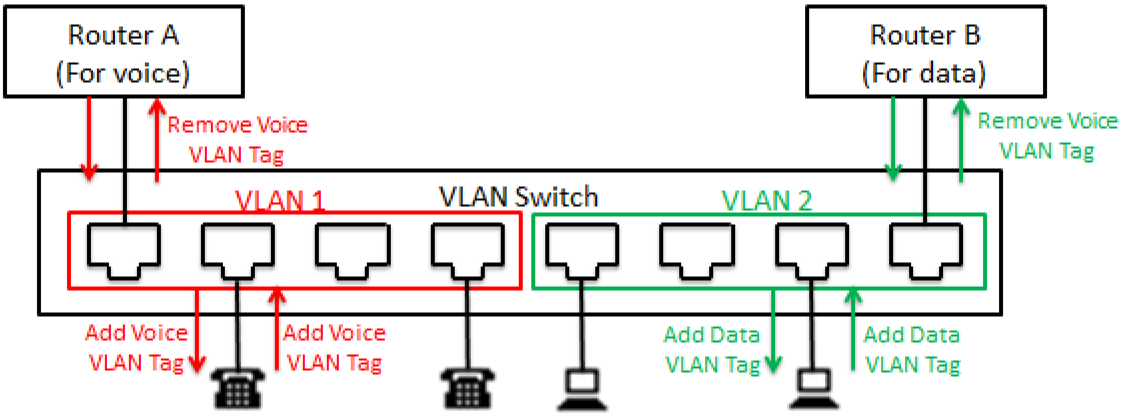
It is a common practice in a managed network to configure voice and data application using different VLANs for prioritization. To separate voice and data traffic by VLAN, the administrator should configure a VLAN switch with two separated VLANs – One for the voice which goes through a high priority trunk while the other for data which goes through a normal priority trunk.

The VLAN switch configuration is different from each VLAN switch provider. The administrator should follow its manual to configure the switch correctly.

For some advanced L2 switch with ALG function, it can identify voice related packets and tagged VLAN ID automatically which needs no configuration on the phone. For basic L2 switch which supports VLAN configuration, it must be configured to have certain ports that belong to the voice VLAN and certain ports that belong to data port and the connected device should have corresponded to tagged VLAN ID, otherwise, the packets will be discarded.

The folowing pictures show two simple configurations on a VLAN switch to support voice and data priority.

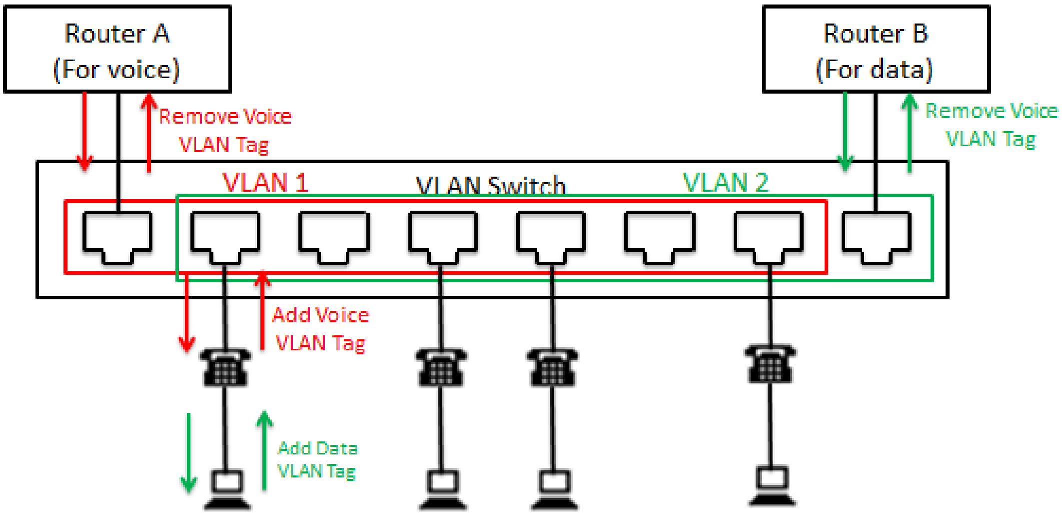
Configure Separated Voice and Data VLAN



VLAN 1 is configured for voice transmission plane. The connected devices should be configured with particular VLAN ID and which will be removed when the packets leave VLAN switch.

VLAN 2 is configured for data transmission plane. For data traffic, it can be configured with data VLAN ID or no VLAN ID.

Configure Overlapped Voice and Data VLAN



3.1.3 VLAN CoS (Class of Service)

VLAN CoS is the IEEE802.1p standard with added traffic class expediting and dynamic multicast filtering to the IEEE 802.1D standard. The CoS value is to specify the priority queue used for specified class if it is supported by the switch. The administrator can assign a different CoS value for media (RTP) and signal (SIP) packets.

The priority level and value are described in the table below:

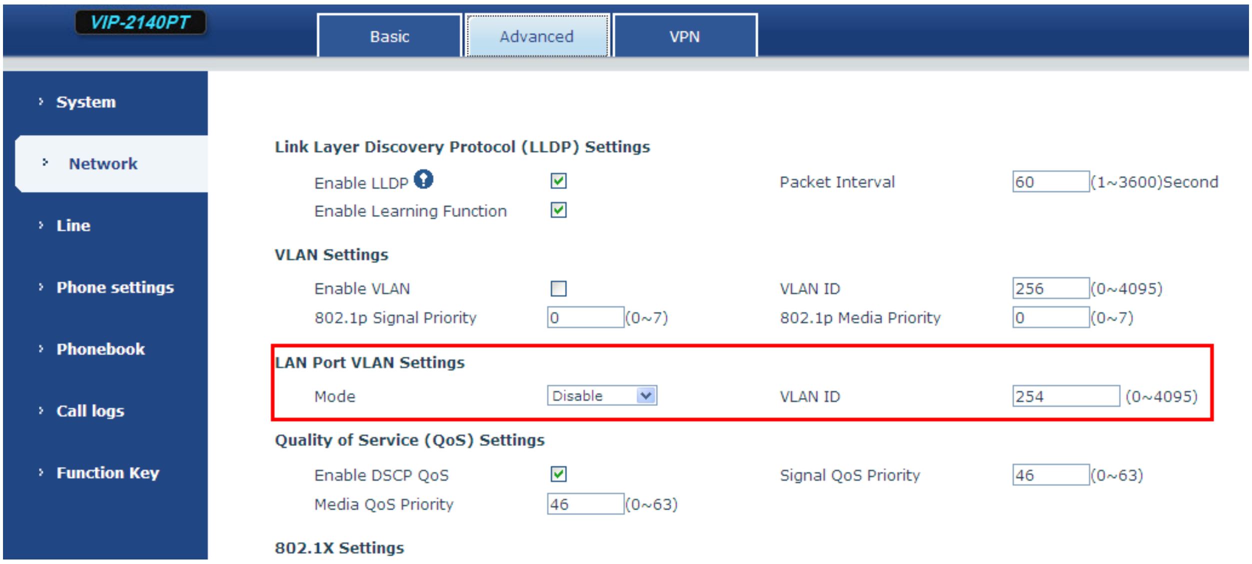
Priority	Acronym	Traffic Types
0 (lowest)	BK	Background
1	BE	Best Effort
2	EE	Excellent Effort
3	CA	Critical Applications
4	VI	Video, < 100 ms latency and jitter
5	VO	Voice, < 10 ms latency and jitter
6	IC	Internetwork Control
7 (highest)	NC	Network Control

3.2 LLDP

The Link Layer Discovery Protocol (LLDP) is a vendor-neutral link layer protocol in the Internet Protocol Suite used by network devices for advertising their identity, capabilities, and neighbors on an IEEE 802 local area network, principally wired Ethernet.

For switches which support LLDP advertising, it can also configure VLAN ID to the connected host.

Both the LAN port and the WAN port of the color screen phones support VLAN settings to authenticate or differentiate the network. Oppositely, the common phone cannot.



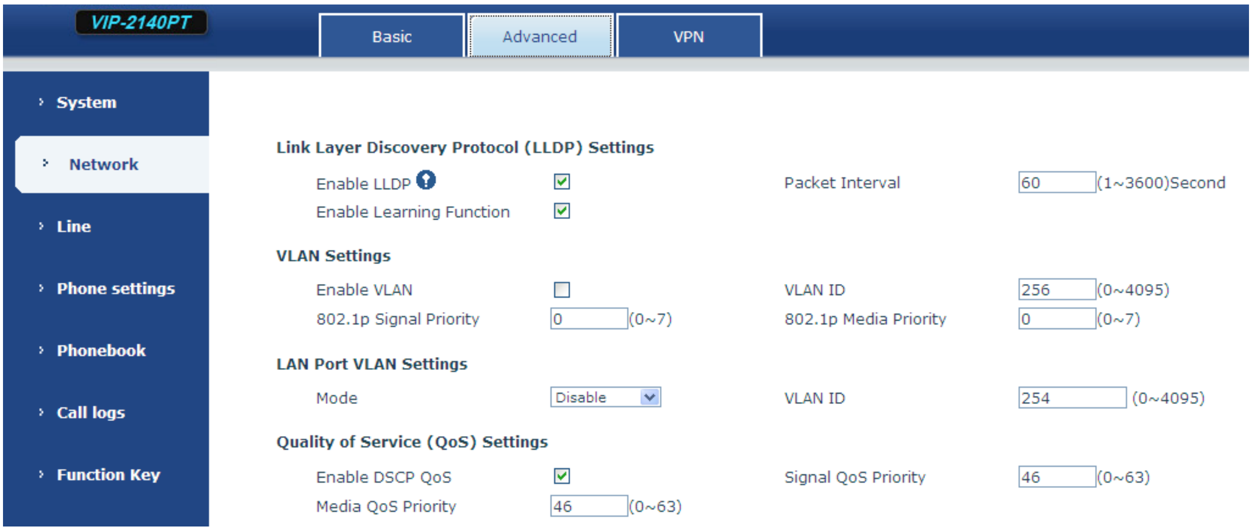
To enable LLDP and VLAN learning, user can check the “Enable LLDP” and “Enable Learning Function” in VIP-1120PT/VIP-2140PT web portal [Network/Advanced /LLDP Settings]

3.3 DSCP

Differentiated services or DiffServ is a computer networking architecture that specifies a simple, scalable and coarse-grained mechanism for classifying and managing network traffic and providing quality of service (QoS) on modern IP networks. DiffServ can, for example, be used to provide low-latency to critical network traffic such as voice or streaming media while providing simple but best service to non-critical services such as web traffic or file transfers.

DiffServ uses a 6-bit differentiated services code point (DSCP) in the 8-bit Differentiated services Field (DS field) in the IP header for packet classification purposes. The DS field and ECN field replace the outdated IPv4 TOS field.

To configure the QoS, user can get access to VIP-1120PT/VIP-2140PT web portal [Network/Advanced / Quality of Service (QoS) Settings]


VIP-2140PT
Basic
Advanced
VPN
System
Network
Line
Phone settings
Phonebook
Call logs
Function Key
Link Layer Discovery Protocol (LLDP) Settings
Enable LLDP
Packet Interval
60
(1~3600)Second
Enable Learning Function
VLAN Settings
Enable VLAN
VLAN ID
256
(0~4095)
802.1p Signal Priority
0
(0~7)
802.1p Media Priority
0
(0~7)
LAN Port VLAN Settings
Mode
Disable
VLAN ID
254
(0~4095)
Quality of Service (QoS) Settings
Enable DSCP QoS
Signal QoS Priority
46
(0~63)
Media QoS Priority
46
(0~63)

CHAPTER 4. ACTION URL AND ACTIVE URI – COLLABORATIVE WORKING ENVIRONMENT THROUGH COMPUTER TELEPHONY INTEGRATION (CTI)

4.1 Introduction

Action URL allows devices to interact with web server applications by sending an HTTP GET request. You can specify a URL that triggers a GET request when a specified event occurs. Action URL can only be triggered by the pre-defined events (e.g., Incoming Call). The valid URL format is:
http://<server address>/<processing file>?<variable name=$variable>.

The HTTP GET request may contain variable name and variable value, separated by "=". Each variable value starts with $ in the query part of the URL. Variable name can be customized by users, while the variable value is pre-defined.
For example, an URL "http://192.168.1.100/newcall.xml?num=$call_id" is specified for the New call, $call_id will be dynamically replaced with the real value of device when the device makes a call.

The following table lists the pre-defined events for action URL.

Event	Description
Setup Completed	When the device completes startup.
Registration Succeeded	When the device successfully registers an account.
Registration Disabled	When the device logs off the registered account.
Registration Failed	When the device fails to register an account.
Phone Off Hooked	When the device is off hook.
Phone On Hooked	When the device is on hook.

Event	Description
Incoming Call	When the device receives an incoming call.
Outgoing Call	When the device places a call.
Call Established	When the device establishes a call.
Call Terminated	When the device terminates a call.
DND Enabled	When the device enables the DND mode.
DND Disabled	When the device disables the DND mode.
Unconditional Call Forward Enabled	When the device enables the unconditional call forward.
Unconditional Call Forward Disabled	When the device disables the unconditional call forward.
Call Forward on Busy Enabled	When the device enables the Call Forward on busy.
Call Forward on Busy Disabled	When the device disables the Call Forward on busy.
Call Forward on No Answer Enabled	When the device enables the Call Forward on no answer.
Call Forward on No Answer Disabled	When the device disables the Call Forward on no answer.
Call transfer	When the device transfers a call.
Unattended Call Transfer	When the device blind transfers a call.
Attended Call Transfer	When the device performs the semi-attended/attended transfer.
Call hold	When the device places a call on hold.
Call resume	When the device retrieves a hold call.
Mute	When the device mutes a call.
UnMute	When the device un-mutes a call.
Missed Call	When the device misses a call.
IP Changed	When the IP address of the device changes.
Idle To Busy	When the state of the device changes from idle to busy.
Busy To Idle	When the state of phone changes from busy to idle.

The following table lists pre-defined variable values.

Variable Value	Description
$Mac	The MAC address of the device
$IP	The IP address of the device
$model	The device model
$firmware	The firmware version of the device
$active URL	The SIP URI of the current account when the device places a call, receives an incoming call or establishes a call.
$active user	The user part of the SIP URI for the current account when the device places a call, receives an incoming call or establishes a call.
$active host	The host part of the SIP URI for the current account when the device places a call, receives an incoming call or establishes a call.
$local host	The SIP URI of the caller when the device places a call. The SIP URI of the callee when the device receives an incoming call.
$remote	The SIP URI of the callee when the device places a call. The SIP URI of the caller when the device receives an incoming call.
$display local	The display name of the caller when the device places a call. The display name of the callee when the device receives an incoming call.
$display remote	The display name of the callee when the device places a call. The display name of the caller when the device receives an incoming call.
$call ID.	The call-id of the active call.

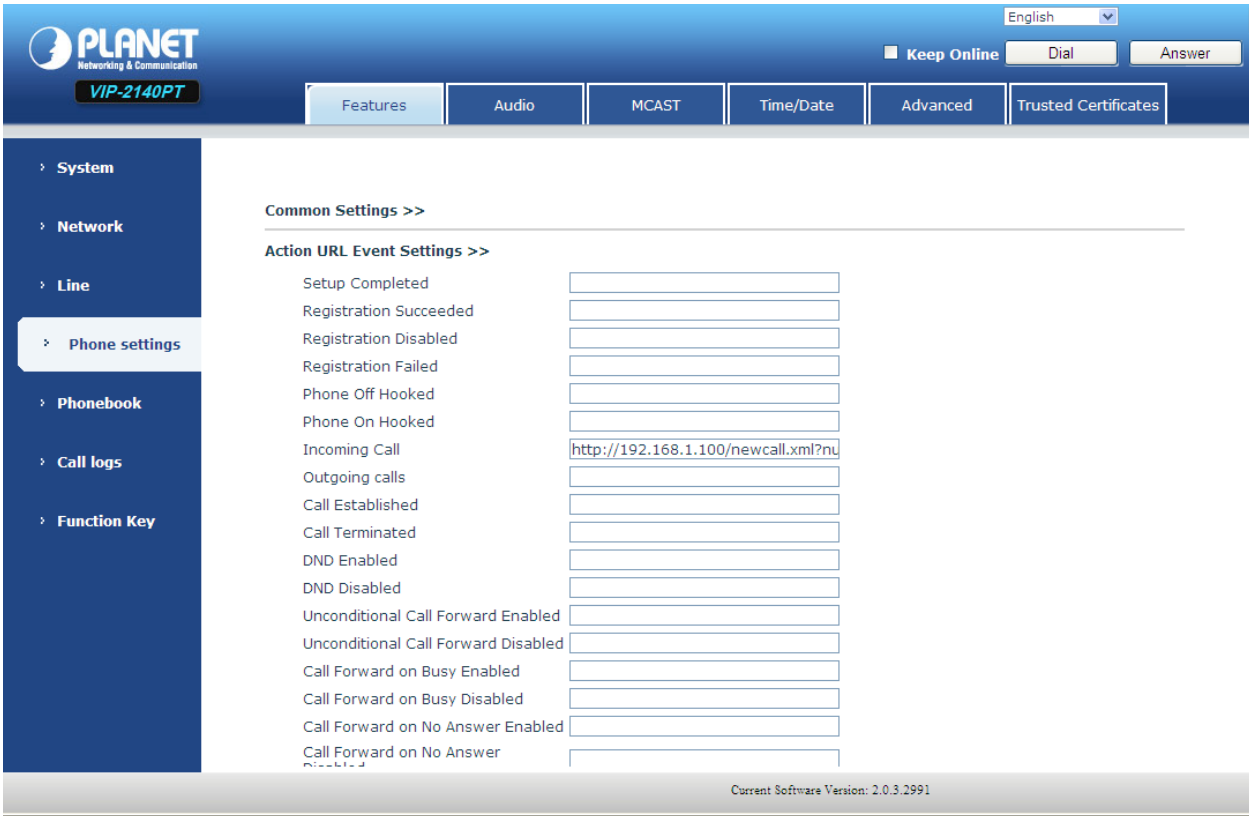
4.2 Action URL configuration

To configure action URL via web user interface:

1 Click on Phone Settings -> -> Features -> Action URL event Settings

2 Enter the action URLs in the corresponding fields.

3 Click Apply to accept the change.



CHAPTER 5.ACTIVE URI

5.1 Introduction

Unlike action URL, active URI allows devices to interact with web server application by receiving and handling a HTTP GET request. When receiving a GET request, the IP phone will perform the specified action and respond with a 200 OK message. A GET request may contain variable named as “key” and variable value, which are separated by “=”. Besides, the command may contain more than one variable, separated by “;”.

The valid URI format is:

http:// <username>:<password>@<server address>/cgi-bin/ConfigManApp.com?key=<variable value>.

For example:

(http://admin:admin@192.168.10.101/cgi-bin/ConfigManApp.com?key=SPEAKER;8311;ENTER)

The following table lists pre-defined variable values:

Variable Value	Description
OK	Press the OK key .
ENTER	Press the Enter soft key.
SPEAKER	Press the Speakerphone key.
RELEASE	Return to the standby status.
F_TRANSFER	Transfers a call to another party.
VOLUME_UP	Increase the volume.
VOLUME_DOWN	Decrease the volume.
MUTE	Mute the call.
F CFWD	Call Forward settings menu
F_HOLD	Place an active call on hold.
F _ CANCEL / X	Cancel actions or reject incoming calls
0-9/*/POUND	Press the keypad (0-9, * or #).
D1-D40	Press the memory keys.
F_CONFERENCE	Press the CONF key or the Conference soft key
F1-F4	Press the soft keys
MSG	Press the MESSAGE key.
F_PBOOK	Press the Phonebook key.

HEADSET / F_HEADSET	Press the HEADSET key.
RD	Press the RD key.
UP/DOWN/LEFT/RIGHT	Press the navigation keys.
Reboot	Reboot the IP phone.
DNDOn	Activate the DND feature.
DNDOff	Deactivate the DND feature.
F_LOCK	Keyboard Lock settings.

5.2 Active URI configuration

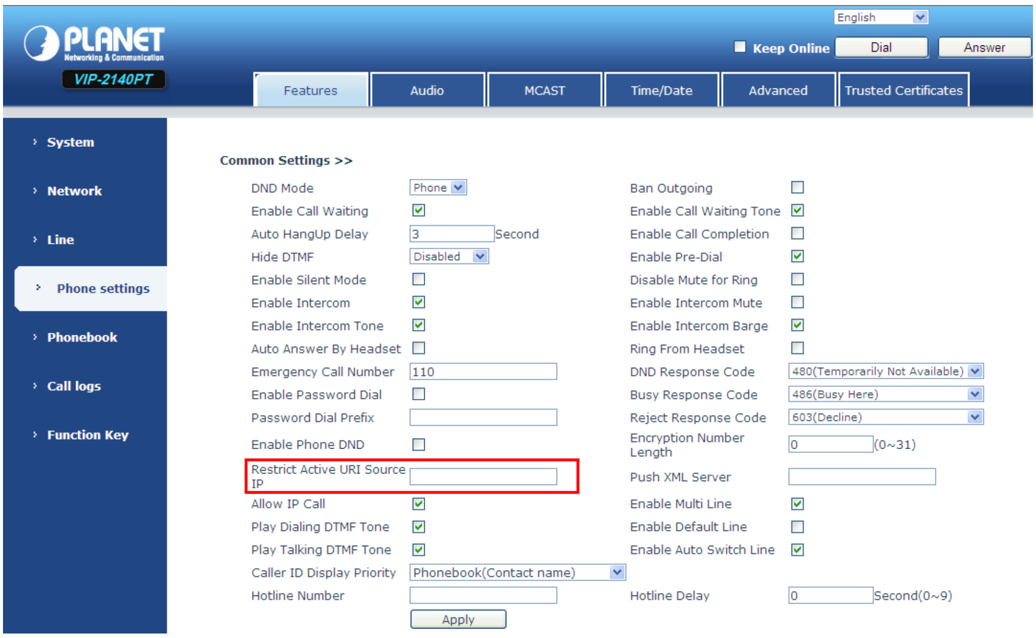
Generally, if someone who is able to get access to the Web server with authentication, the devices will receive all the HTTP GET requests by default. But for security reasons, on some cases, users want to limit restriction to active URI Source IP. You need to specify the trusted IP address for Active URI. You can specify a trusted IP address on the IP phone, or configure the IP phone to receive and handle the URI from any IP address.

To configure action URL via web user interface:

1 Click on Phone Settings -> Features -> Restrict Active URI Source IP

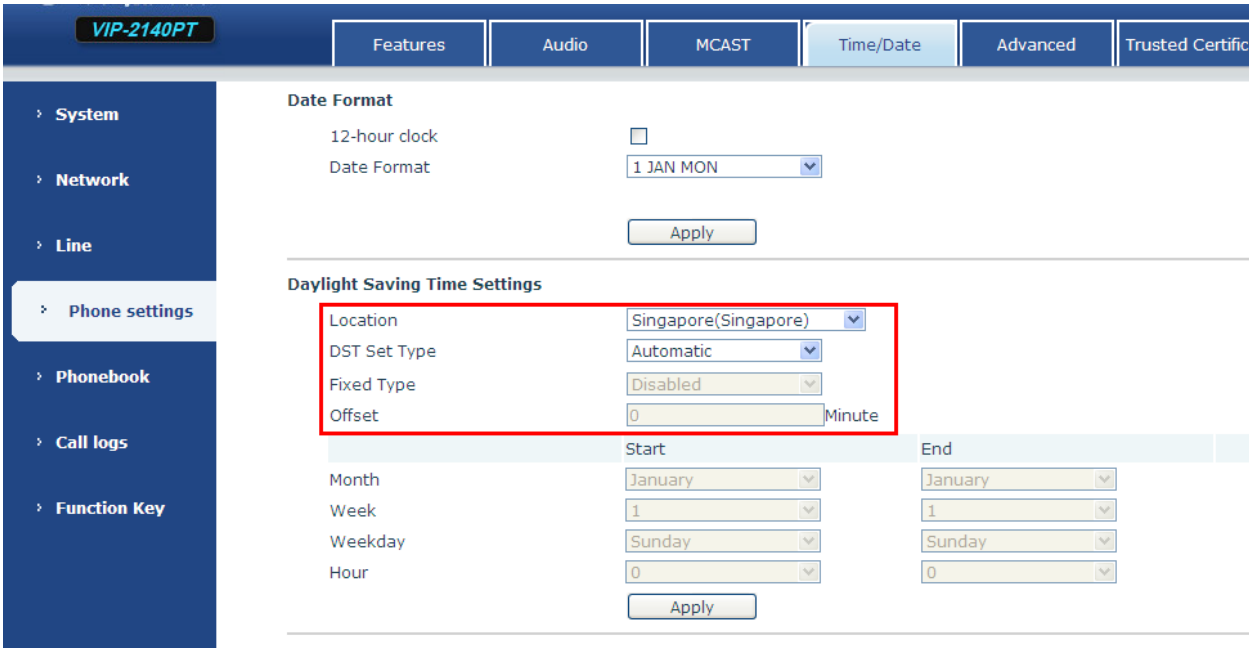
2 Enter the IP address or any in the Active URI that allows IP. If you leave the field blank, the IP phone will receive or handle any HTTP GET request with authentication.

3 Click Apply to accept the change.



5.3 Daylight Saving Time Settings

The color screen product time zone configuration is different with common screen of the VIP-1120PT/VIP-2140PT product. The color screen product have the Daylight Saving time Settings. You can set it via Phone Settings—>Time/Date—>Daylight Saving Time Settings as the following dialog shows:



CHAPTER 6. TRAVERSING NAT

6.1 Introduction

Network Address Translation (NAT) is a methodology of remapping internal IP address space to external one by modifying source/destination network address and port field in IP packets when they are transmitted through router or firewall. Not all of network devices could be assigned to a public IP address when it accesses internet due to the shortage of amount of IPv4 address. The NAT helps a lot of these network devices to share public IP address to visit Internet.

But it is very hard to send information to the device under NAT. And it is crucial to VoIP device because server needs to know how to send incoming call request to the phone in local private network. NAT traversal has been proposed to solve this problem. There are many technologies that could be used to do NAT Transversal, such as STUN (RFC5389), ICE(RFC5245), TURN(RFC5766), symmetric Response (RFC3581, Rport), etc. In this guide, we only discuss two NAT Traversal methods which X series phone have implemented.

6.2 STUN

Session Traversal Utilities for NAT (STUN) is a network protocol to allow network device to discover its public IP address through the assistant of 3-party sever on public network. It could also discover the NAT type and binding port of NAT for local network device.

After applying the appropriate configurations for STUN, or loading the STUN configuration, our X series phone would access STUN server first to detect the public IP address, NAT binding port and type of NAT. There are four types of NATs generally, which STUN could not handle the traversal under symmetric NAT. After the STUN server responds successfully, phone would send packets to STUN server through SIP local port (default 5060), and wait the response to modify SIP URI. Then a request from local network to public SIP server would contain URI information like packets to which a public IP address sent. Sever feeds back response to the public IP address and port would come back to local device through NAT. So, server could find local device under NAT easily. When phone call is made, it would use STUN to discover binding RTP port on NAT again, and fill out the information on SDP session. And then a voice media connection would be built between local device and public device.

Note:

1. STUN could not handle the case under symmetric NAT.

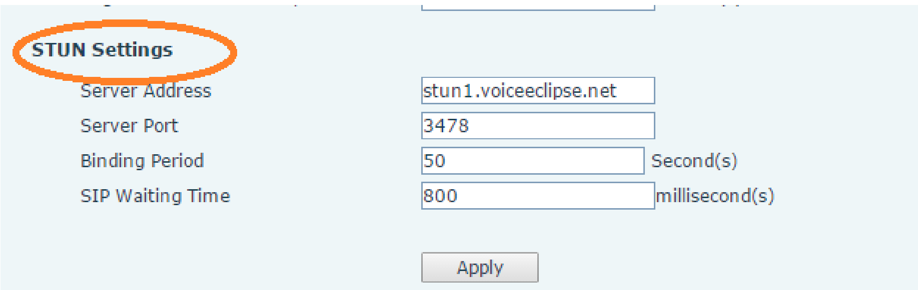
2. In some special situations, phones under same local network could not set up connection successfully when STUN is enabled. Because some NATs would refuse to transmit data packets on two binding ports over NAT. So default setting of STUN is disabled. STUN could be enabled by web user interface or importing configuration file.

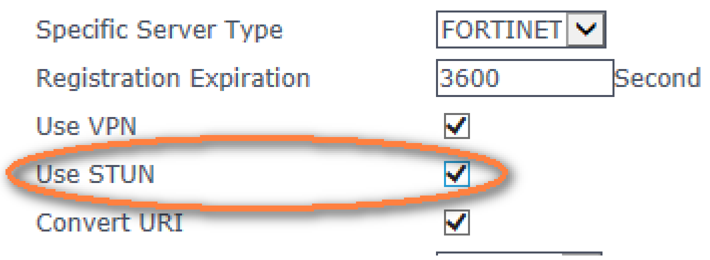
Configuration method:

1. Web user interface

1) Click ‘Line/Basic Settings ’.
2) Modify the field ‘STUN Settings/ Server Address’ with domain name of STUN sever or IP address.
3) Click the “Apply” button to save the change.
4) Click ‘Line/SIP/Advanced Settings’ when Line 1 needs to use STUN to do NAT Traversal.
5) Make sure that ‘use STUN’ checkbox is checked.
6) Click the ‘Apply’ button to save the change.

Here is an example:





2. Items in configuration file

There are also two parts in configuration file for STUN, common configuration and line configuration.

1) Common configuration

Parameters	Descriptions
STUN Server :	Configure Domain or IP address of STUN server. Default value is empty.
STUN Port :	Configure STUN server port, default is 3478。
STUN Refresh Time :	Configure STUN request sending period by

	seconds. Default value is 50 seconds.
SIP Wait Stun Time :	Configure the timeout which phone would wait STUN response before SIP request is sent. Default value is 800 ms.

Here is an example to set STUN server to 'stun1.voiceeclipse.net'.

a) Export configuration files from Web user interface.

b) Find '<SIP CONFIG MODULE>'

c) Modify the fields shown below:

STUN Server :stun1.voiceeclipse.net

STUN Port :3478

STUN Refresh Time :50

SIP Wait Stun Time :800

2) Line configuration:

Parameters	Descriptions
SIP1 NAT Type :	Configure Line 1 that uses STUN as its NAT traversal method. 0 is Disabled, 1 is Enabled. Default is 0.
SIP2 NAT Type :	Configure Line 2 that uses STUN as its NAT traversal method.

Note: These two configuration items should be listed below ' --SIP Line List-- :'

Here is example to set line 1 STUN enable and line 2 disable.

--SIP Line List-- :

SIP1 NAT Type :1

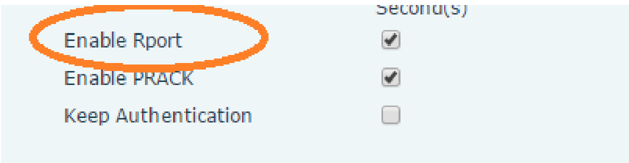
SIP2 NAT Type :0

6.3 Rport

Rport is another NAT Traversal method in SIP stacks. It is also a keyword string in SIP information. RFC3581 describes the implementation of Rport, which is also called Symmetric Response. After SIP server received a UDP request, it should check whether there is 'rport' string in the 'Via' field and its value is empty. If so, the server should insert 'received' parameter and 'rport' parameter in the 'via' field when it sends response to the client. The value of 'received' parameter should be the IP address from which its packets are directly. And the address might be NAT device's address or phone's address connecting to public network. In addition, the value of 'rport' parameter is equalled to port from public IP address. SIP server will keep client device's NAT mapping IP and port information and send request via the field. This method would not need 3-party server to find NAT mapping information, and have been supported in many types of SIP server. So the default configuration value of Rport in X series phone is 1 (enabled).

Configuration method:

1. configuration by web user interface
    1) Click 'Line/SIP/Advanced Settings' when you want to configure Rport setting for Line1
    2) Make sure checkbox of 'Enable Rport' is checked when you want Rport to be active.
    3) Click the 'Apply' button to save the changes.



2. Items in configuration file

Parameters	Descriptions
SIP1 Enable Rport	Configure Line 1 enables Rport feature or not. 0 is Disabled, 1 is Enabled. Default is 1.