D-Link®

DPH-80 IP Phone

User's Guide

Warranty and Registration

Trademarks

Copyright Statement

Hardware:

D-Link warrants each of its hardware products to be free from defects in workmanship and materials under normal use and service for a period commencing on the date of purchase from D-Link or its Authorized Reseller and extending for the length of time stipulated by the Authorized Reseller or D-Link Branch Office nearest to the place of purchase.

This Warranty applies on the condition that the product Registration Card is filled out and returned to a D-Link office within ninety (90) days of purchase. A list of D-Link offices is provided at the back of this manual, together with a copy of the Registration Card.

If the product proves defective within the applicable warranty period, D-Link will provide repair or replacement of the product. D-Link shall have the sole discretion whether to repair or replace, and replacement product may be new or reconditioned. Replacement product shall be of equivalent or better specifications, relative to the defective product, but need not be identical. Any product or part repaired by D-Link pursuant to this warranty shall have a warranty period of not less than 90 days, from date of such repair, irrespective of any earlier expiration of original warranty period. When D-Link provides replacement, then the defective product becomes the property of D-Link.

Warranty service may be obtained by contacting a D-Link office within the applicable warranty period, and requesting a Return Material Authorization (RMA) number. If a Registration Card for the product in question has not been returned to D-Link, then a proof of purchase (such as a copy of the dated purchase invoice) must be provided. If Purchaser's circumstances require special handling of warranty correction, then at the time of requesting RMA number, Purchaser may also propose special procedure as may be suitable to the case.

After an RMA number is issued, the defective product must be packaged securely in the original or other suitable shipping package to ensure that it will not be damaged in transit, and the RMA number must be prominently marked on the outside of the package. The package must be mailed or otherwise shipped to D-Link with all costs of mailing/shipping/insurance prepaid. D-Link shall never be responsible for any software, firmware, information, or memory data of Purchaser contained in, stored on, or integrated with any product returned to D-Link pursuant to this warranty.

Any package returned to D-Link without an RMA number will be rejected and shipped back to Purchaser at Purchaser's expense, and D-Link reserves the right in such a case to levy a reasonable handling charge in addition mailing or shipping costs.

Software:

Warranty service for software products may be obtained by contacting a D-Link office within the applicable warranty period. A list of D-Link offices is provided at the back of this manual, together with a copy of the Registration Card. If a Registration Card for the product in question has not been returned to a D-Link office, then a proof of purchase (such as a copy of the dated purchase invoice) must be provided when requesting warranty service. The term "purchase" in this software warranty refers to the purchase transaction and resulting license to use such software.

D-Link warrants that its software products will perform in substantial conformance with the applicable product documentation provided by D-Link with such software product, for a period of ninety (90) days from the date of purchase from D-Link or its Authorized Reseller. D-Link warrants the magnetic media, on which D-Link provides its software product, against failure during the same warranty period. This warranty applies to purchased software, and to replacement software provided by D-Link pursuant to this warranty, but shall not apply to any update or replacement which may be provided for download via the Internet, or to any update which may otherwise be provided free of charge.

D-Link's sole obligation under this software warranty shall be to replace any defective software product with product which substantially conforms to D-Link's applicable product documentation. Purchaser assumes responsibility for the selection of appropriate application and system/platform software and associated reference materials. D-Link makes no warranty that its software products will work in combination with any hardware, or any application or system/platform software product provided by any third party, excepting only such products as are expressly represented, in D-Link's applicable product documentation as being compatible. D-Link's obligation under this warranty shall be a reasonable effort to provide compatibility, but D-Link shall have no obligation to provide compatibility when there is fault in the third-party hardware or software. D-Link makes no warranty that operation of its software products will be uninterrupted or absolutely error-free, and no warranty that all defects in the software product, within or without the scope of D-Link's applicable product documentation, will be corrected.

D-Link Offices for Registration and Warranty Service

The product's Registration Card, provided at the back of this manual, must be sent to a D-Link office. To obtain an RMA number for warranty service as to a hardware product, or to obtain warranty service as to a software product, contact the D-Link office nearest you. An addresses/telephone/fax list of D-Link offices is provided in the back of this manual.

LIMITATION OF WARRANTIES

IF THE D-LINK PRODUCT DOES NOT OPERATE AS WARRANTED ABOVE, THE CUSTOMER'S SOLE REMEDY SHALL BE, AT D-LINK'S OPTION, REPAIR OR REPLACEMENT. THE FOREGOING WARRANTIES AND REMEDIES ARE EXCLUSIVE AND ARE IN LIEU OF ALL OTHER WARRANTIES, EXPRESSED OR IMPLIED, EITHER IN FACT OR BY OPERATION OF LAW, STATUTORY OR OTHERWISE, INCLUDING WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. D-LINK NEITHER ASSUMES NOR AUTHORIZES ANY OTHER PERSON TO ASSUME FOR IT ANY OTHER LIABILITY IN CONNECTION WITH THE SALE, INSTALLATION MAINTENANCE OR USE OF D-LINK'S PRODUCTS

D-LINK SHALL NOT BE LIABLE UNDER THIS WARRANTY IF ITS TESTING AND EXAMINATION DISCLOSE THAT THE ALLEGED DEFECT IN THE PRODUCT DOES NOT EXIST OR WAS CAUSED BY THE CUSTOMER'S OR ANY THIRD PERSON'S MISUSE, NEGLECT, IMPROPER INSTALLATION OR TESTING, UNAUTHORIZED ATTEMPTS TO REPAIR, OR ANY OTHER CAUSE BEYOND THE RANGE OF THE INTENDED USE, OR BY ACCIDENT, FIRE, LIGHTNING OR OTHER HAZARD.

LIMITATION OF LIABILITY

IN NO EVENT WILL D-LINK BE LIABLE FOR ANY DAMAGES, INCLUDING LOSS OF DATA, LOSS OF PROFITS, COST OF COVER OR OTHER INCIDENTAL, CONSEQUENTIAL OR INDIRECT DAMAGES ARISING OUT THE INSTALLATION, MAINTENANCE, USE, PERFORMANCE, FAILURE OR INTERRUPTION OF A D-LINK PRODUCT, HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY. THIS LIMITATION WILL APPLY EVEN IF D-LINK HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

IF YOU PURCHASED A D-LINK PRODUCT IN THE UNITED STATES, SOME STATES DO NOT ALLOW THE LIMITATION OR EXCLUSION OF LIABILITY FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATION MAY NOT APPLY TO YOU.

FCC Warning

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

– Reorient or relocate the receiving antenna.

– Increase the separation between the equipment and receiver.

– Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

– Consult the dealer or an experienced radio/TV technician for help.

The device complies with part 15 of the FCC rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference.

(2) This device must accept any interference received, including interference that may cause undesired operation.

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Introduction

The D-Link DPH-80 is a fully featured Ethernet business phone that allows both business and residential customers to benefit from IP Telephony services. It reduces costs by receiving local and long distance voice services and data services over a single network connection.

This easy-to-use IP Phone simply plugs right into the local area network through a standard RJ45 interface. The DPH-80 utilizes 10/100BASE-TX for Ethernet connectivity and supports telephone network features such as Call Redial. In addition, it provides access to a host of features for business applications, including hold, mute and one-touch dialing.

Features

- IP address assignment using DHCP (Dynamic Host Configuration Protocol) or static configuration
- Silence suppression and comfort noise generation that make the user feel natural when connected
- Adaptive jitter buffer for smooth voice reception
- DTMF tone generation
- Lost packet recovery ability for improved voice quality
- Adjustable speaker/ringer volume control
- Remote software update support
- Easy to use set-up and programming interface
- One-touch dialing (*Note:* this feature is not a speakerphone. It allows you to dial a number without using the handset. Once the party you are calling picks up, you must use the handset to talk.)
- Call hold
- Last number redial

Overview

The D-Link DPH-80 is a low cost, simple to use and extremely versatile IP phone with the look and feel of a normal PSTN phone.

The D-Link IP Phone can operate under any of the three main Internet telephony protocols: Media Gateway Control Protocol (MGCP); Session Initiation Protocol (SIP); and the H.323 protocol. These protocols are used for signaling, maintaining, and tearing down voice calls. The D-Link IP phone allows voice data to be carried over the same path used by your computer for the Internet or Local Area Network (LAN).

The D-Link DPH-80 phone is easy to install and supports plug and play features of the IP network. Out of the box, your IP phone will work in any of the above-mentioned three protocol infrastructures with minimal configuration. Advanced, customized configuration is easily achieved through a web-browser configuration utility.

D-Link IP phones support remote maintenance, allowing software to be upgraded remotely for new features and any bug fixes. The DPH-80 supports a unique remote diagnostic feature to monitor phone functions and performance.

The DPH-80 implements a sophisticated low-latency adaptive jitter buffer algorithm to combat packet loss and jitter in inter-arrival time of voice transmissions.

This User Manual describes the features and functions of the D-Link IP Phone, with separate sections for MGCP, SIP, and H.323 configuration and troubleshooting. This manual along with release notes should help anyone to install and use the D-Link MGCP phone without support from the developers.

Installation

Unpacking

Open the shipping carton and carefully remove all items. In addition to this User's Guide, make sure you have received all of the following items:

- IP phone
- Handset
- Handset cord
- Power adapter

If any item appears to be missing or damaged, please contact your local reseller.

System Requirements

- Internet connection (via ISP)
- Local power outlet

Installation Procedure

The following are steps to install and power-on your DPH-80 IP phone.

1. Connect the RJ-45 Ethernet cable from the DPH-80 to a LAN jack.
2. Plug the power adapter into the appropriate wall outlet.
3. Plug the power adapter plug into the power jack.

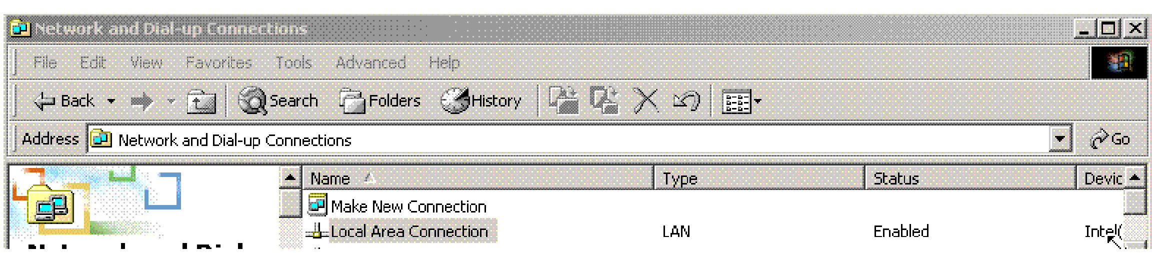
Basic Configuration

IP Address

In order to use a Web browser to configure the DPH-80 IP phone, you must make sure the phone has a valid Ethernet connection to a PC or LAN via its Ethernet port. We recommend using a recent version of any widely used browser such as Netscape or Internet Explorer. The browser must have JavaScript enabled.

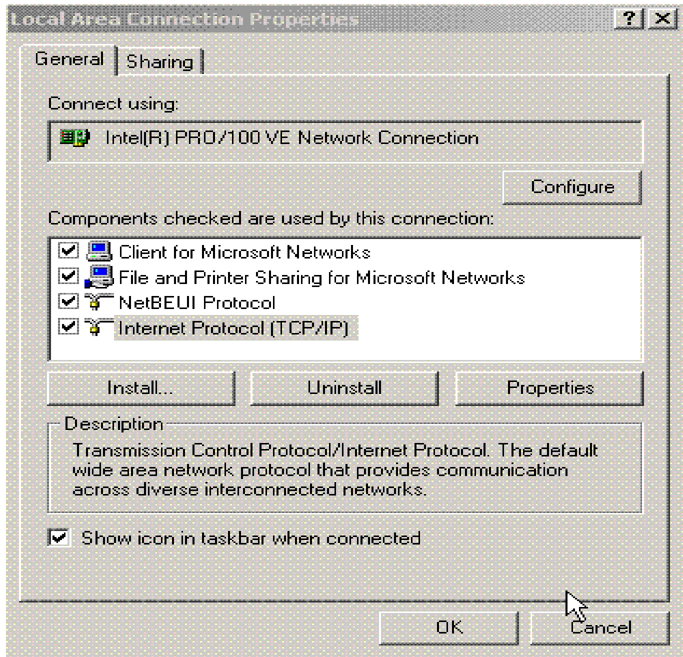
The DPH-80 comes with a default IP address of 10.1.1.80. You must make sure the PC is in the same IP domain as the IP phone. You can do this by changing the IP address of the PC as shown below.

- In Windows, go to **Start/Settings/Control Panel/Network and Dial-Up Connections**.

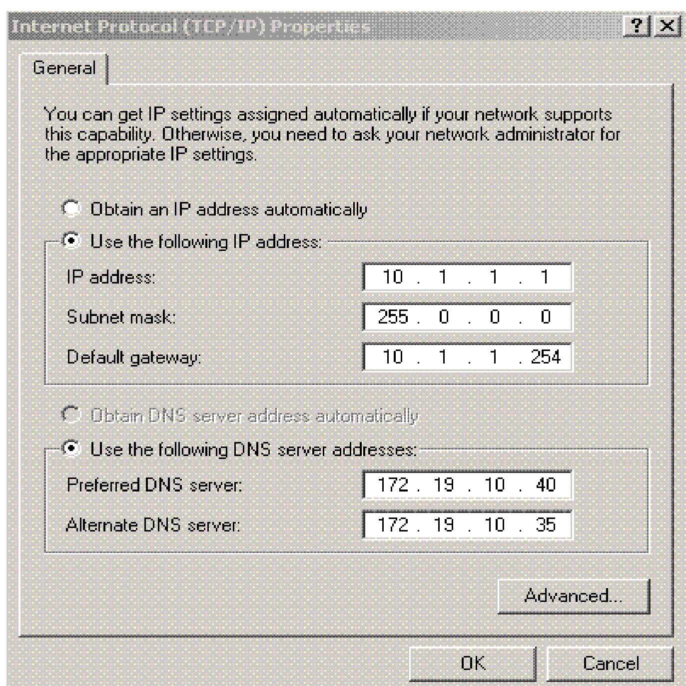


Right-click on Local Area Connection (LAN).

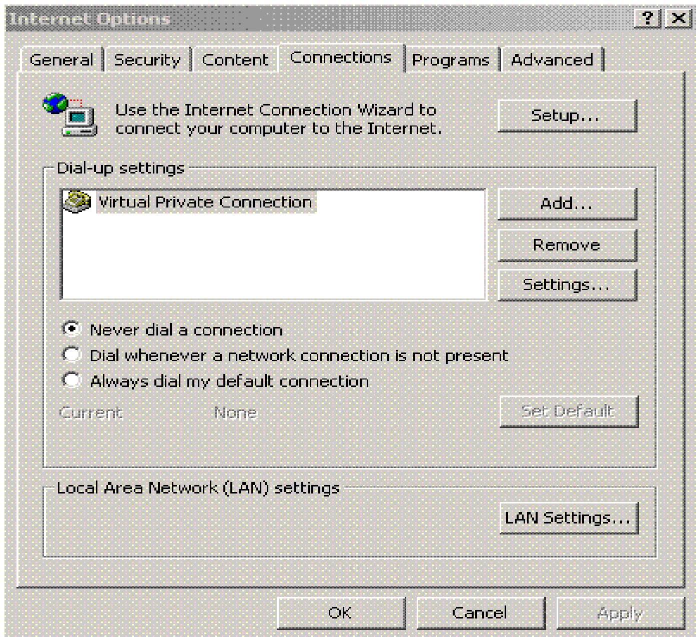
Click on **Properties.**



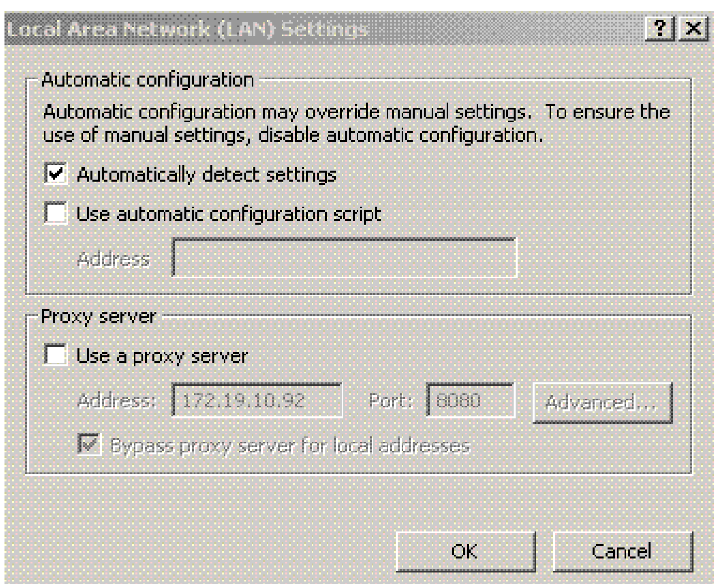
- In the General Tab, click on **Internet Protocol (TCP/IP).**



- Click on **Use the following IP address**, and enter a value in the 10.1.1.xx range. (Do not use the IP address 10.1.1.80; that address is already in use by the DPH-80 as a default address.) If necessary, change the Subnet mask and the Default gateway to match the values shown above.
- Click **OK.**
- Open Internet Explorer. Click on **Tools/Internet Options/Connections**



- Click on **LAN Settings.**



- Make sure **Use a proxy server** is disabled.
- Click **OK.**

Final Configuration

The DPH-80 is highly versatile and can be configured to operate in any of the three main Internet telephony protocols – MCGP, SIP, and H.323.

Please choose the appropriate protocol and follow the directions indicated in the Web-based configuration utility.

Appendix_A	MGCP (Media Gate Control Protocol)
Appendix B	SIP (Session Initiation Protocol)
Appendix C	H.323 Protocol

Each appendix contains configuration details, instructions for use, and a troubleshooting guide.

Appendix A

DPH-80 MGCP (Media Gate Control Protocol) IP Phone Configuration

Infrastructure Requirements

Though the DPH-80 MGCP phone will work in any type of LAN network, a 100mbps, switched network is more suitable for providing good quality voice communications.

MGCP phones need a Media Gateway Controller or Call Agent or Notified Entity.

To operate properly, the DPH-80 needs a set of IP parameters such as IP address, subnet mask, gateway address, and DNS server address. These parameters can be configured either statically through a browser or dynamically through DHCP or PPPoE. A DHCP server in the local LAN is required to provide these parameters.

The D-Link MGCP phone has many configurable parameters. These parameters can be configured through any Java-enabled Internet browser (Netscape 6.2 or above, IE 5.0 or above).

If your LAN network has a firewall and NAT, they should support MGCP to make and receive calls from outside your LAN network.

A TFTP server is required to support remote software upgrading. The TFTP server should have the two software image files (dph80v1.tfp and dph80v2.tfp) from D-Link in the current directory of the TFTP server selected by ‘set path’.

Configuring the MGCP phone

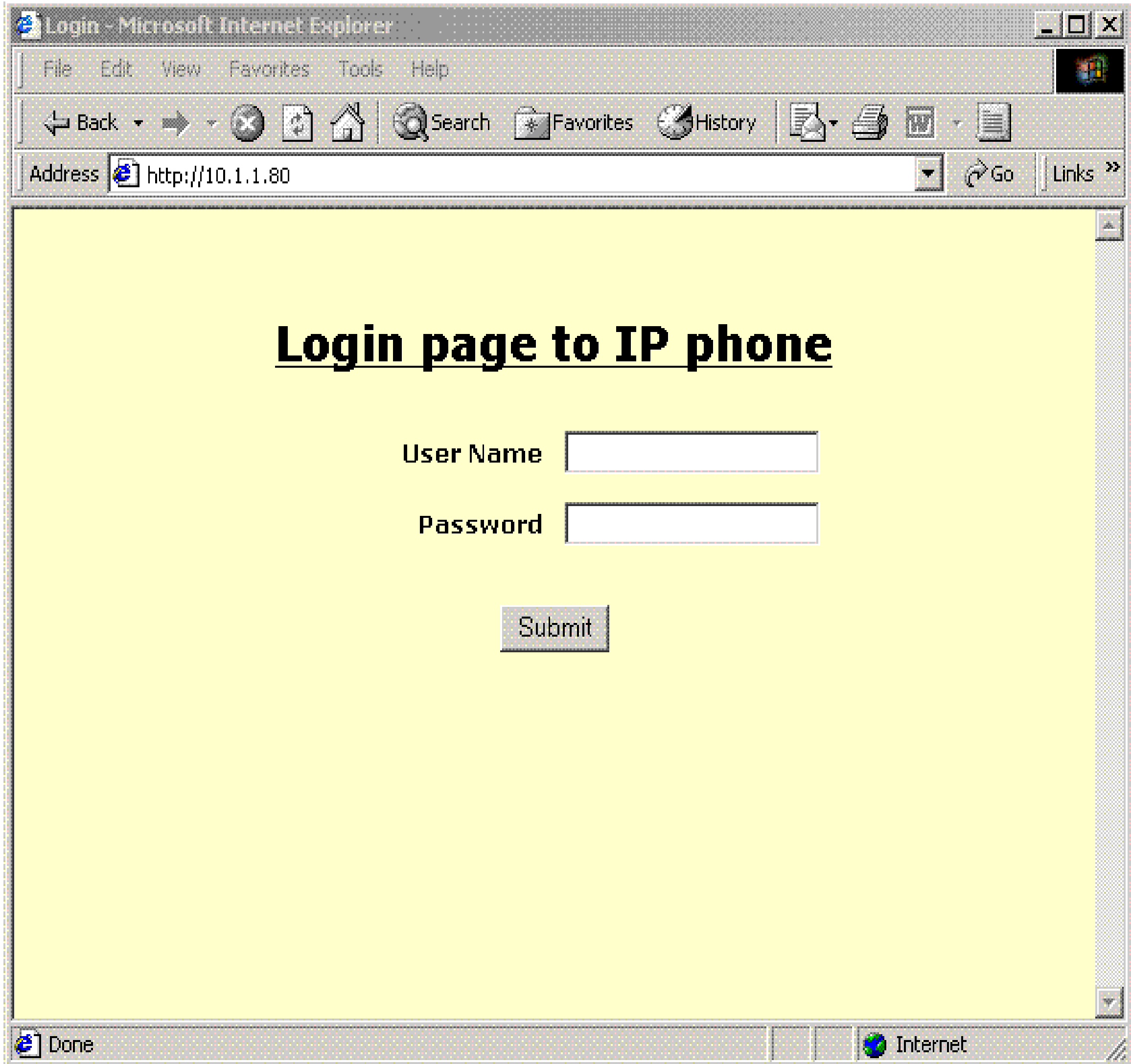
Once you have the above infrastructure in place, you can power up the MGCP phone. The MGCP phone will play *till it gets the response from Media Gateway Controller*. If it does not get the dial tone within the expected time, the MGCP phone is not configured. However, the MGCP phone is accessible through an Internet browser for configuration.

To access the web interface for the D-Link DPH-80:

Use a JavaScript-enabled Internet browser (Netscape 6.2 or above, IE 5.0 or above) with the default IP address of the DPH-80 entered in the address box (http://10.1.1.80).

- The following page will appear.

Login Page to IP phone



The following two parameters control access to MGCP phone. Default value for both will be "dlink". These values can be changed later using the *Change Login Name and Password* Page.

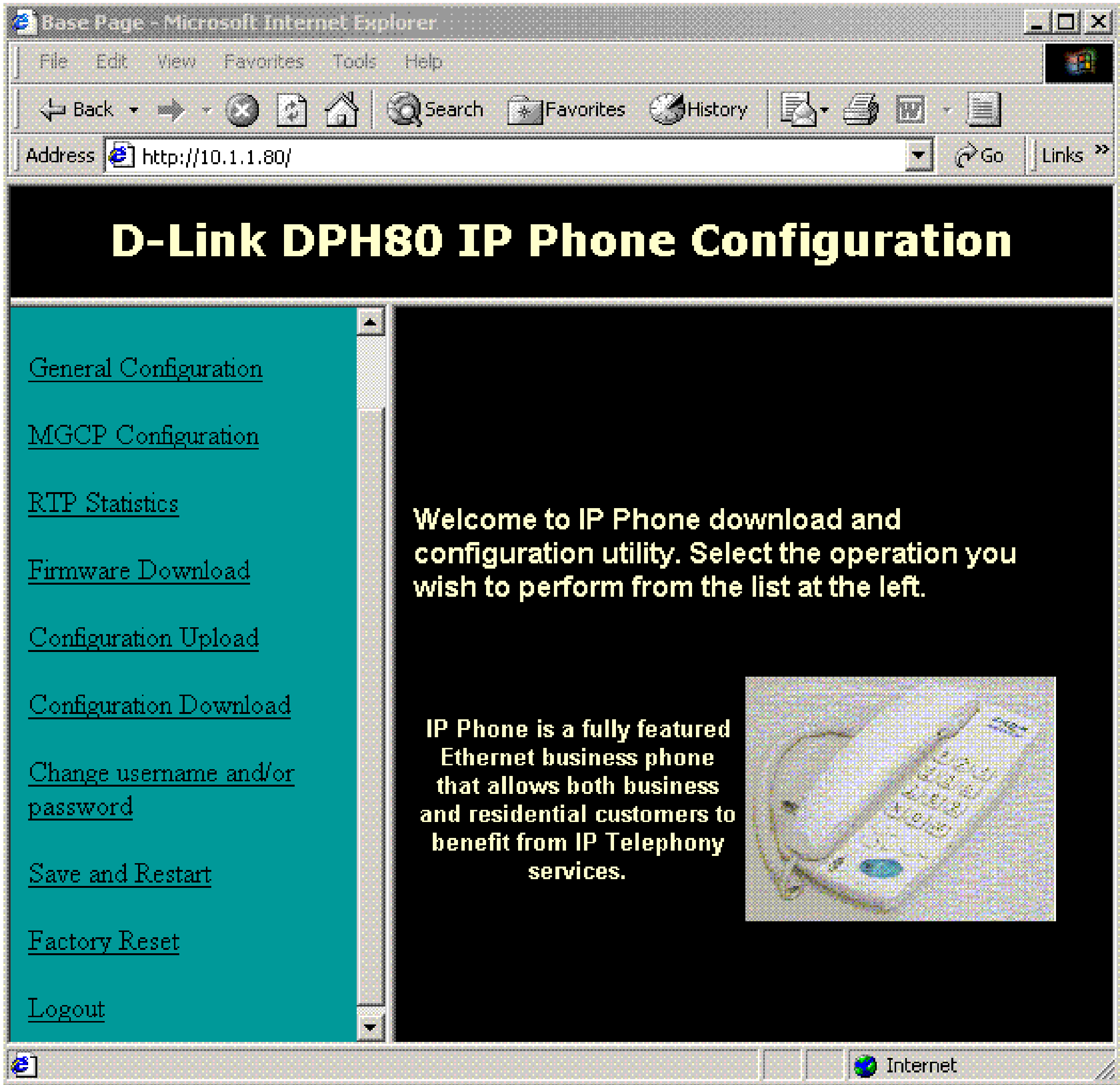
User Name: This is case-insensitive with a maximum of 20 characters.

Password: This is case-sensitive with a maximum of 20 characters.

Click **Submit.**

After logging in the *D-Link DPH-80 phone configuration* page is displayed and this page provides links to other pages.

Configuration Main Page



Click on General Configuration. A new page containing information about your system and the DPH-80 will appear.

General Configuration

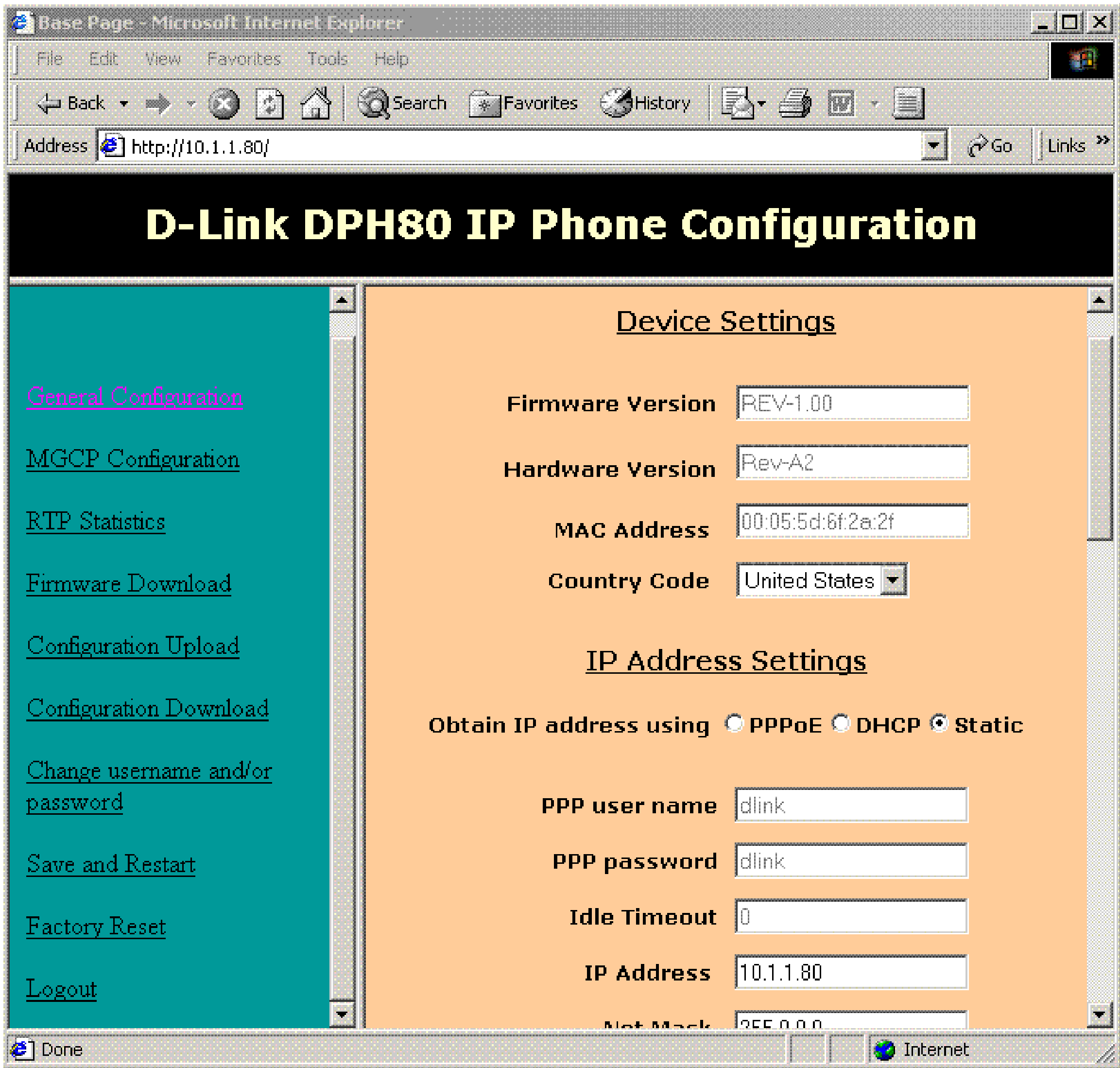
Firmware Version : It shows the current firmware version of the IP phone. It is updated whenever the MGCP phone software is updated. It cannot be modified.

Hardware Version: It shows the current hardware version of the IP phone. It cannot be modified.

MAC Address: This shows the MAC address of the board in colon-separated hex form. Initially it contains the default value ff:ff:ff:ff:ff:ff. It can be modified once to a non-default value. *Once modified, it will be grayed out and cannot be changed again.*

Country Code: This is a drop-down menu. Select the appropriate country. This field controls the type of tones played by MGCP phone.

Obtain IP address using: If *static* option is selected, then a user-configured IP address, Net Mask, Default gateway, and DNS server address would be used for the phone. If *DHCP* is



selected, then these values will be obtained using DHCP. If the PPPoE is selected, and using the PPP username and password for authentication, PPPoE obtains an IP address for the phone. Default selection is Static-enabled.

PPP user name: This is the user name used for PPP authentication with the PPP server while obtaining an IP address via PPPoE.

PPP password: This is the password used for PPP authentication with the PPP server while obtaining an IP address via PPPoE.

Idle Timeout: This is the time interval in seconds of session inactivity after which the PPP session should be terminated. If this is set to 0, then the session will never be terminated. This field is grayed out at the moment so that it can't be modified. This will allow the PPP session to be on permanently unless the server closes the connection. This field can be made active later to enable a configuration of timeout value.

IP Address: This should have the IP address of the phone in dot-separated IP address form. An illegal IP address won't be allowed for this field.

Net Mask: This will have the Net Mask of the network to which the IP phone is connected. It must be in dot-separated form. An illegal IP address mask won't be allowed for this field.

Default Gateway: This is the default gateway for the IP phone. An illegal IP address won't be allowed for this field.

DNS server Address: This is the IP address of the DNS server, which will respond to the DNS queries from the IP phone. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

TFTP Server: This has the IP address of the host where TFTP server is running. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

Firmware Filename (up to 6 characters): This is the filename which you want to download from the TFTP server. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Upload Filename (up to 6 characters): This is the filename to upload the configuration parameters from the phone to the TFTP server. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Download Filename (up to 6 characters): This is the filename to download the configuration parameters from the TFTP server to the phone. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Adaptive Jitter: If this is enabled then Jitter Buffer will be adaptive, otherwise it will use a fixed buffer of a size specified in *Maximum Buffer Size.*

Maximum Buffer Size: If adaptive jitter is disabled, the phone will use this static value for Jitter Buffer size. This should be in the range of 0-300 in ms.

Log Server: This flag is turned on in case the user wants to log all debug messages for viewing.

Log Server Address: This has the IP address of the machine where all the log messages should be sent. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

Log Server Port: This is the port number on the log server to which the log messages are to be sent. It should be a valid port number in the range of 0-65335. The user should make sure that it is not one of the reserved port numbers.

Microphone Gain: This will show the microphone gain in the range of -14 to 14 (unit of dB)

Speaker Gain: This will show the speaker gain in the range of -14 to 14 (unit of dB)

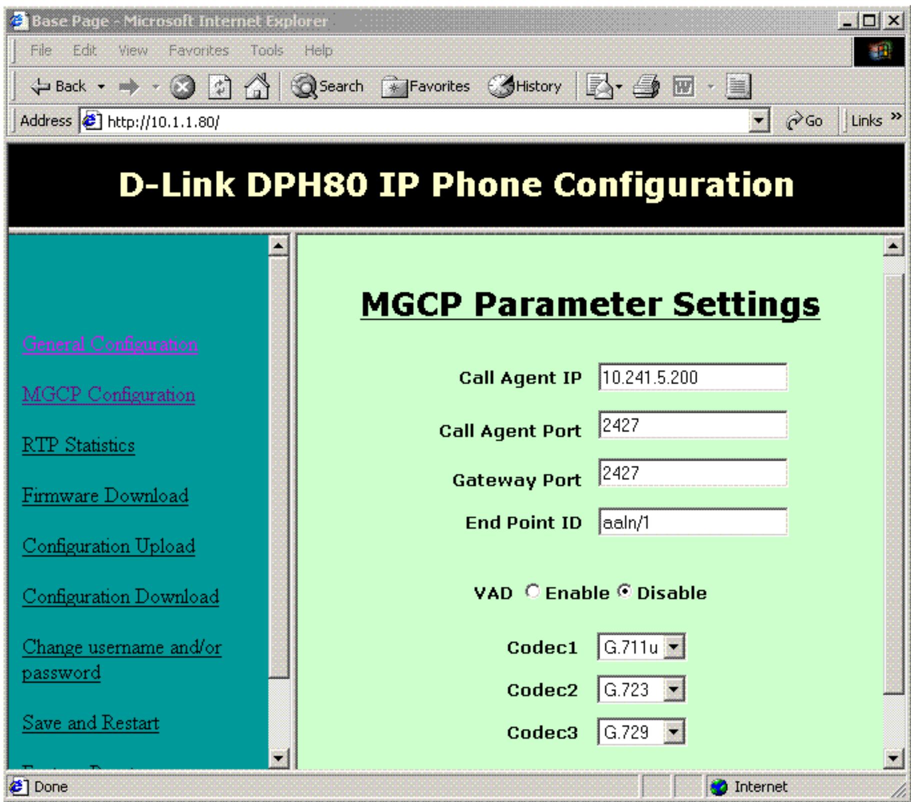
Access Settings: The following three key sequences should be unique.

Factory Default: This is the key sequence the user should dial on the phone to get the phone to use all the default values of the parameters. After entering this key sequence on the MGCP phone it will restore the parameters to default upon next restart.

Production Key: This is the key sequence the user should dial on the phone to get to production-test mode. After entering this key sequence, MGCP phone will start in production-test mode upon next restart. Once again if we reboot it will start functioning in the MGCP phone mode.

TFTP upload: This is the key sequence the user should dial on the phone to start the TFTP software update. After getting the new image, the phone will start itself using the new image.

MCGP Configuration



Call Agent IP: It must be in dot-separated form. An illegal IP address won't be allowed for this field. (Ex: 10.241.5.200).

Call Agent Port Number: This is the port at which the Call Agent receives and sends packets.(Ex: 2427).

Gateway Port: This is the port number at which the MGCP Phone will open the socket to send and receive packets. (Ex: 2427).

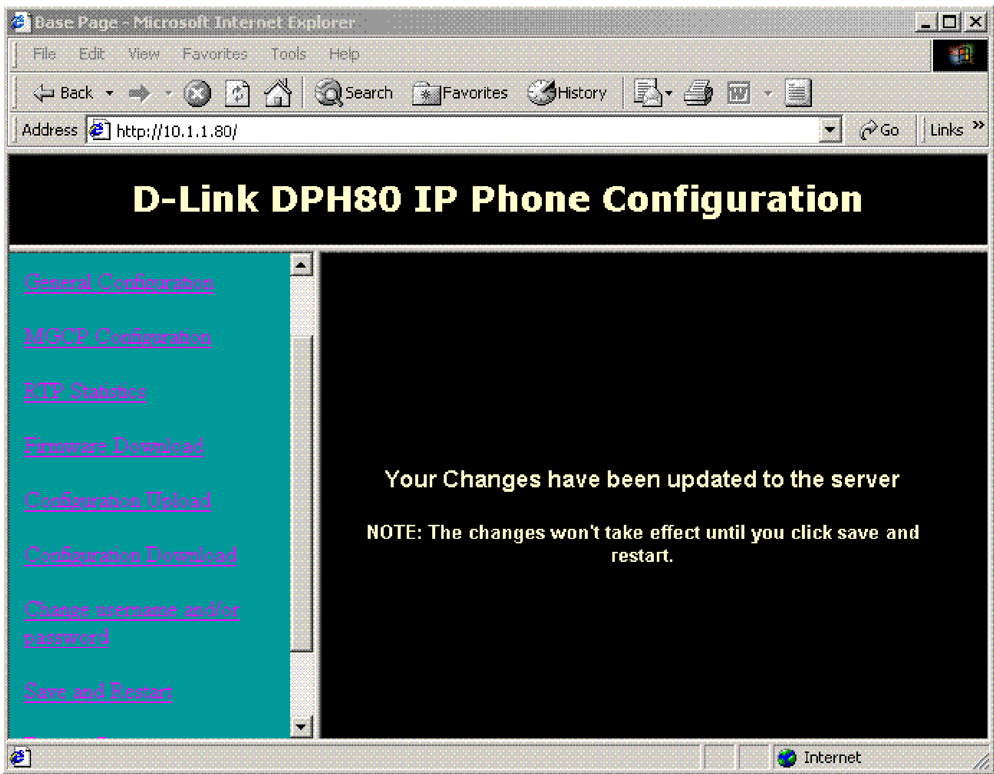
Endpoint ID: This is the endpoint identifier as defined in rfc 2705 (*endpointname@[IP* Address or domain name]). Here we can specify only the endpoint name (up to @) or full identifier with either domain name or IP address. If only the endpoint name (up to @) is given, the identifier will be formed automatically by software by appending the IP address.

- *Example 1*: If the endpoint name is "dlink/1" the identifier will be formed by software by appending IP address like "dlink/1@[10.241.5.231].

- *Example 2:* If the full identifier is given as dlink/1@book. The IP address will not be appended.

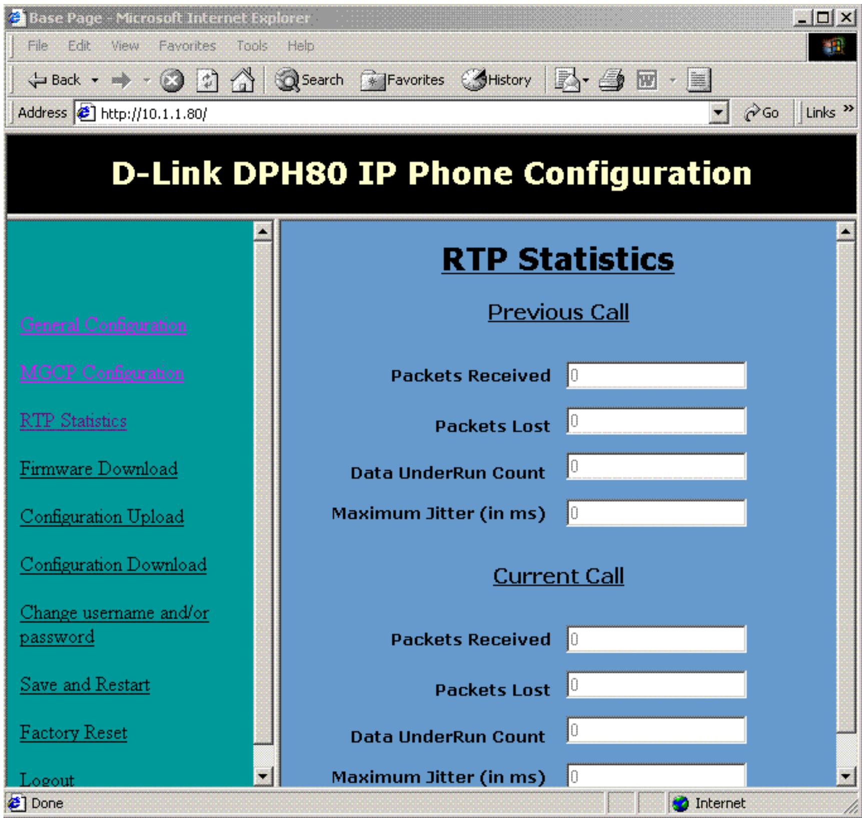
Codec1, Codec2 and *Codec3* are drop-down menus. This allows selecting what codecs are to be used by the phone. It also specifies the priority of the codec while negotiating for the codec to use in any call. Codec1 will be given the highest priority.

After entering the appropriate values, click **Submit**. The following page will appear.



Do not click **Save and Restart** until you have finished configuration.

RTP Statistics



This is an informational page and shows the RTP statistical data from the current call and the previous call. This page is refreshed every 5 seconds automatically.

Packets Received: Number of packets that have been received for the call.

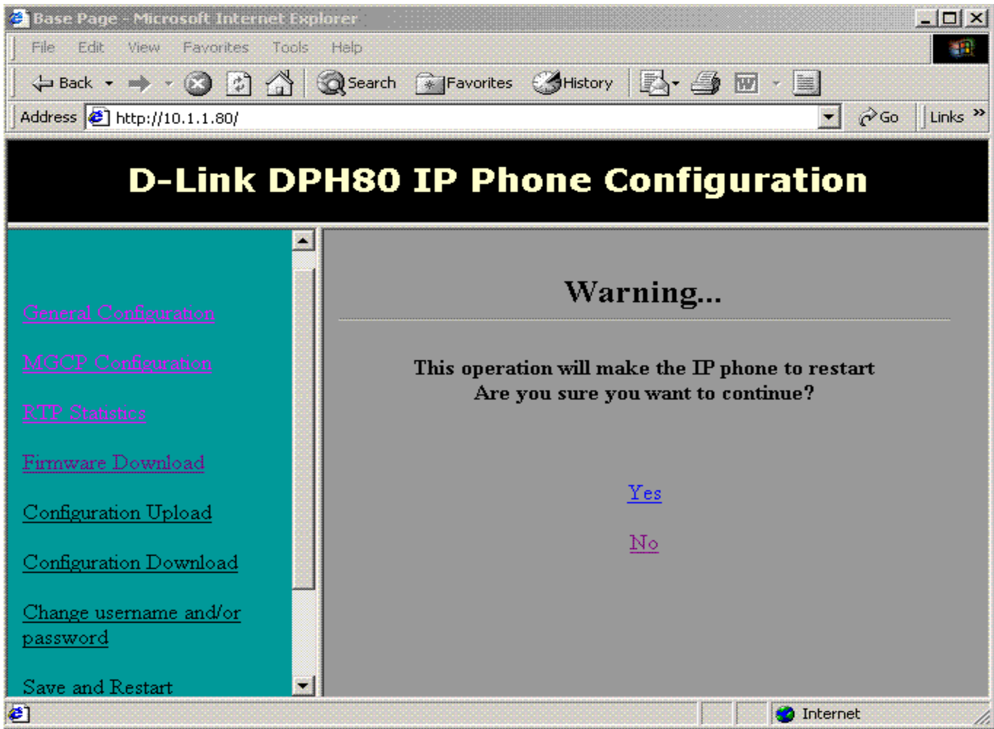
Packets Lost: Number of packets that have been lost in the network.

Data Under Run Count: This is the jitter buffer under run count for the entire call.

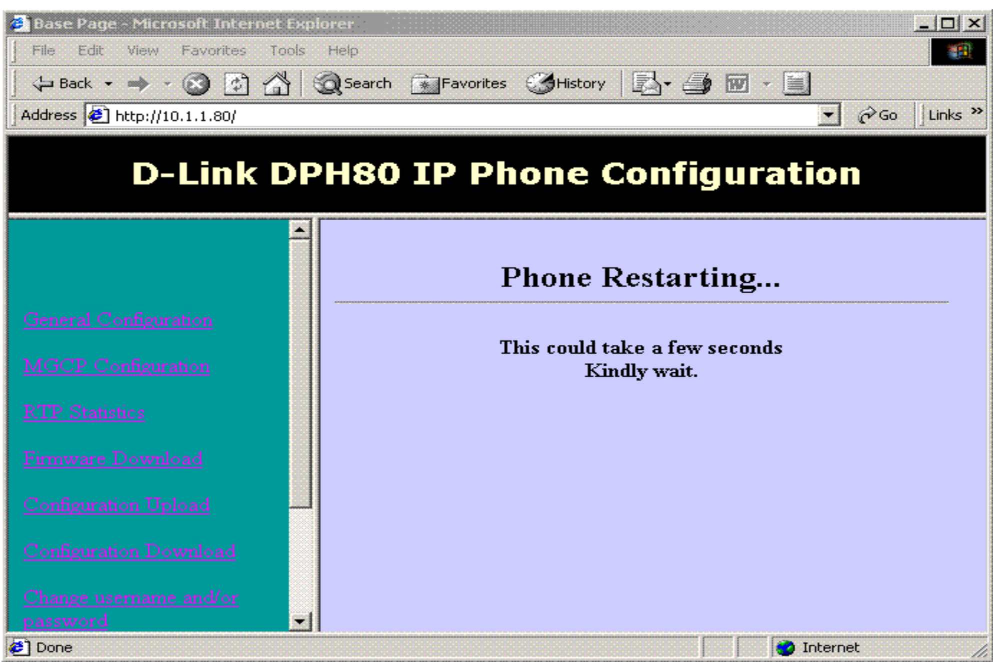
Maximum Jitter: This is the estimated maximum jitter in the network, shown in units of ms.

Firmware Download

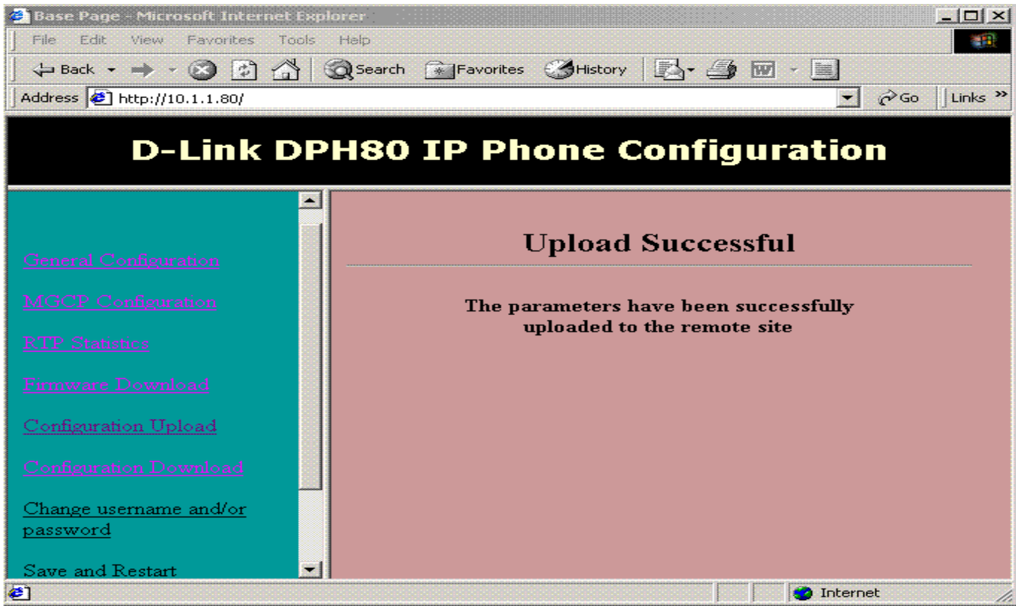
When clicked, this link will display a warning page. Click *'yes'* to download the firmware from the TFTP server to the firmware filename. The TFTP server and filename are set in the General Configuration. Click *'no'* on the warning page to return to the main page.



On clicking Yes the following screen will appear.

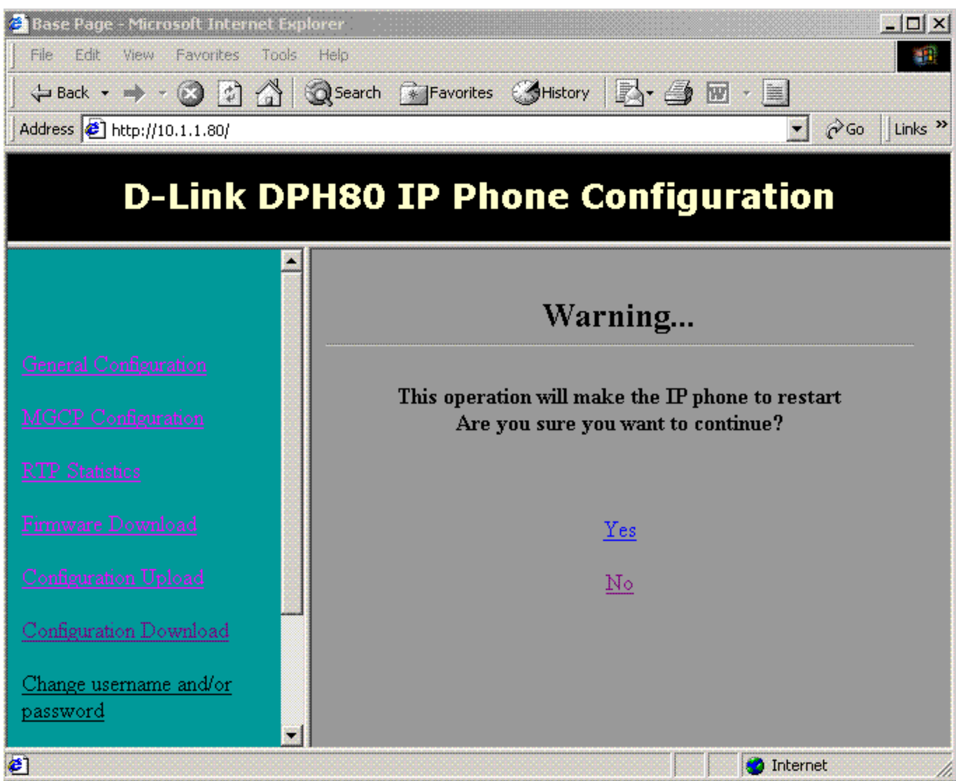


Configuration Upload



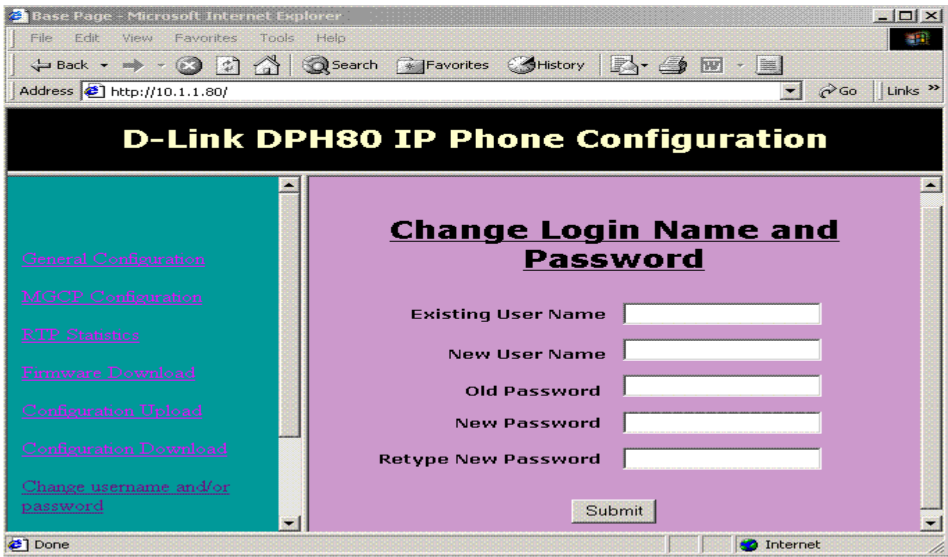
When clicked, this link will display a warning page. Click *'yes'* to upload the configuration parameters from the phone to the TFTP server as the upload filename. The TFTP server and filename are set in the General Configuration. Click '*no'* on the warning page to return to the main page.

Configuration Download



When clicked, this link will display a warning page. Click *'yes'* to download the configuration parameters from the TFTP server to the phone as the download filename. The TFTP server and filename are set in the General Configuration. Click *'no'* on the warning page to return to the main page.

Change Login Name and Password



Existing User Name: This is the user name that was used to access the MGCP phone from the web browser. This is case-insensitive and may be 20 characters long at maximum.

New User Name: If the user wants to change the login user name, it should be entered here. Otherwise, enter the same user name. This is case-insensitive and may be 20 characters long at maximum.

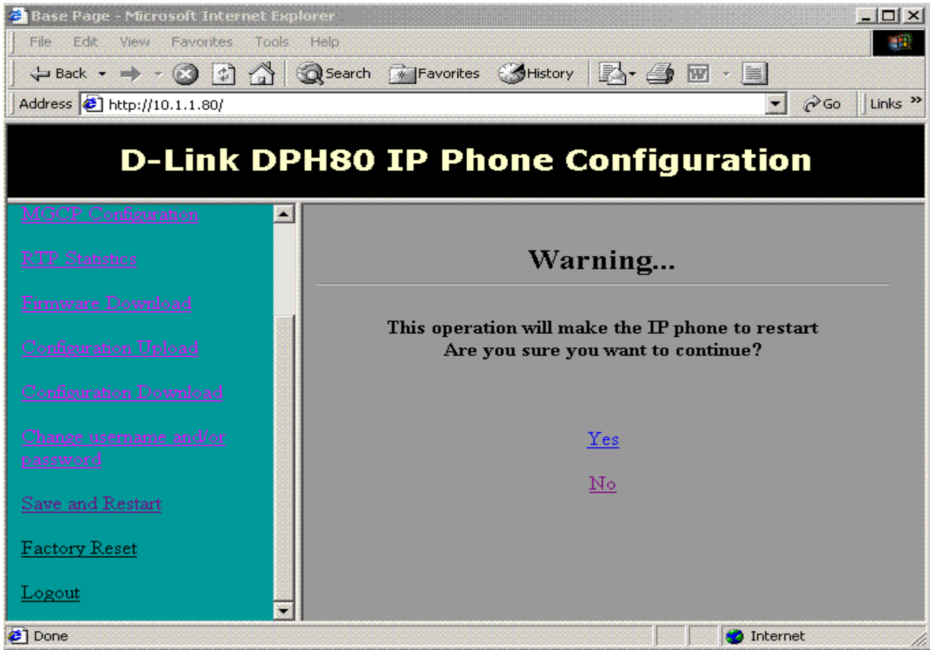
Old Password: This is the login password used to access the MGCP phone from the web browser. This is case-sensitive and may be 20 characters long at maximum.

New Password: A new login password should be entered here. This is case-sensitive and may be 20 characters long at maximum.

Retype New Password: The above field value should be retyped here to confirm that the correct value was written. If the two don't match, the user will be prompted to retype them.

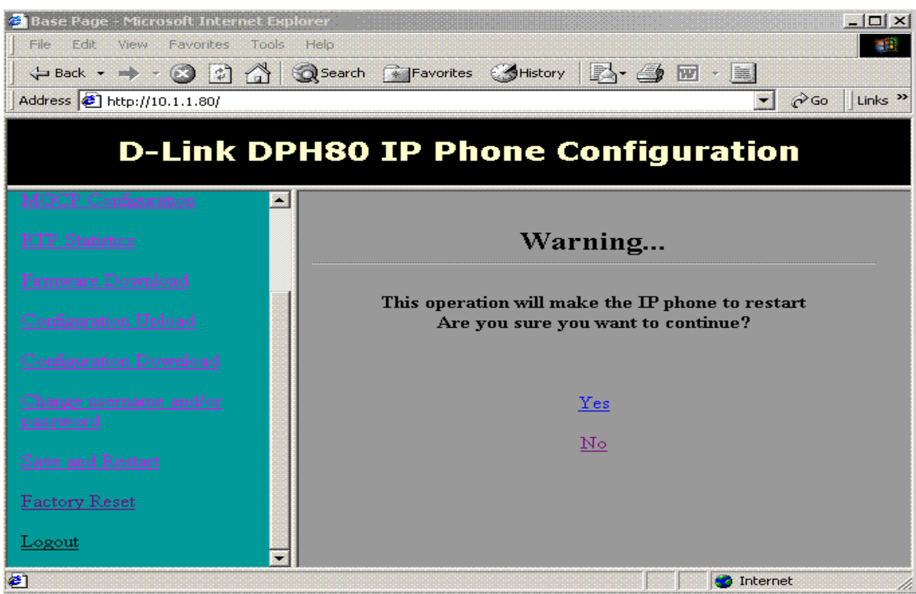
Click Submit.

Save and Restart



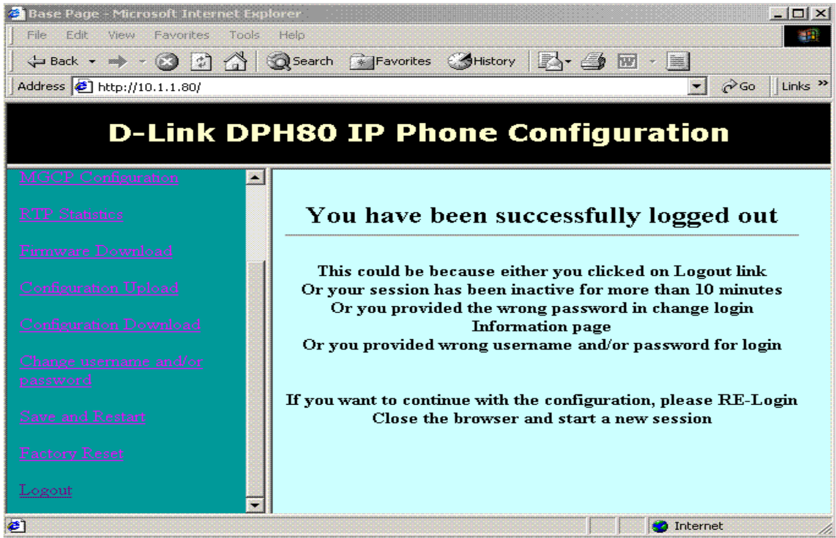
When clicked, this link will display a warning page. Click *'yes'* to save all the updated parameters to the flash memory and restart the phone so that the latest changes take effect. The *You have been successfully logged out* page will be displayed. The phone takes about 30 seconds to come up again. Click *'no'* on the warning page to return to the main page.

Factory Reset



When clicked, this link will display a warning page. Click *'yes'* to reset the phone to factory defaults and automatically restart. Click *'no'* on the warning page to return to the main page.

Logout



When clicked, the *You have been successfully logged out* page is displayed and the current configuration session is terminated.

Note: This page may also be displayed if you provided the wrong username and/or password or if your session has been inactive for more than 10 minutes. If you are having an active session with the server, any other user accessing the MGCP phone's configuration will get the *Server Busy* page and will not be allowed access.

Using the MGCP Phone

If the MGCP phone is configured properly and if the support infrastructure is in place, the MGCP phone will play the dial tone on off-hook. You can dial any registered MGCP number by entering the number in sequence; the end of the number will be automatically detected by using the following two methods:

- Using “Digit map” algorithm with the digit map supplied by Call Agent.
- Inter digit time out (2 seconds).

MGCP Phone Features

D-Link DPH-80 works in 10, 100, and 10/100 Mbps Ethernet environment. It has an adjustable handset and speaker volume control and it plays tone for all numerical key press.

LEDs

- *Link/Activity:* Steady on for link up, flashing for activity, off for link down
- *Speaker LED (Red):* indicates speaker-on status
- *Hold (Green):* Steady on to indicate Hold status; off means normal
- *Mute (Red):* Steady on to indicate Mute status; off means normal

Tones

The DPH-80 MGCP phone plays the following tones depending on the phone’s current state. It supports different type of tones for different countries (selected through configuration).

- Dial tone
- Call progress tone
- Ring back tone
- Busy tone
- Call alert (ringing) tone
- Error tone
- DTMF tones for all numeric keys
- Call Waiting Tone

Calling Features

- MUTE: When pressed, the MGCP phone turns off the microphone signal from the handset but still plays voice from the other party.
- HOLD: When pressed, the MGCP phone disconnects both microphone and speaker while connection is kept alive. No voice packets are transmitted from the D-Link MGCP phone. Hold LED is on. User may press again to release the call. This feature requires support from the remote phone for proper functioning.
- REDIAL: When pressed, redials last dialed number.
- TRANSFER: Toggle the hook-switch quickly to flash (transfer) the call. The MGCP phone plays dial tone. Then enter the party to transfer the call by the general dialing method. The MGCP phone transfers the call and plays busy tone. Flashing the hook twice before dialing the number will restore the call to normal state (to call-active state).
- SPEAKERPHONE: One-touch dialing key. When pressed, speaker LED is on and speaker itself is on while on-hook. If user off-hooks after dialing or presses this button again the one-touch operation is terminated, LED and speaker are both turned off. *Note: This is not a true speakerphone, but is designed to allow one-touch dialing. Although the other party can hear through the speakerphone, voice quality is very poor.*
- CALL WAITING: Call-waiting tone will be played whenever a new party calls while a call is in progress. By pressing the hook-switch MGCP will switch to the incoming call. Pressing again will switch between two parties.

Algorithms

Codecs: D-Link MGCP phones supports G.711 U/A law, G.723.1, and G.729AB. The browser configuration allows selecting codecs and their priority.

Voice activity detection, silence suppression, and comfort noise generation: The VAD can be disabled in the configuration irrespective of the codec being used.

Adaptive Jitter Buffer: D-Link MGCP phones uses a robust adaptive jitter buffer algorithm. It can be disabled and a fixed-size jitter buffer can be used instead through configuration.

Other features

Remote software upgrade: A predefined key sequence will download the MGCP phone software and restart the phone. The MGCP phone should have been configured with the correct TFTP server IP address.

Remote diagnosis: The MGCP phone will send status and other messages to the log server configured in the MGCP phone. The remote log server should run the server application from D-Link to receive and display these messages. This feature can be disabled through the browser.

Restore factory defaults: If you enter the specified key sequence, the MGCP phone restores the configurable parameters to default values upon next restart.

Production testing.

If you enter the specific key sequence, the D-Link MGCP phone will execute a production test upon next restart. The production test is described later in this section.

MGCP Troubleshooting

Error Conditions

The D-Link MGCP phone will detect the following error conditions and play the error tone.

Error tone on network-connection failure: Upon network connection, tone will revert to normal dial tone. The link LED also gives this information.

Error tone if there is no DHCP server: The phone will revert to normal dial tone on detecting a DHCP server.

Error tone if the MGCP proxy is down on power-up: The phone will revert to normal dial tone on detecting an MGCP server.

Error tone if the called party is not registered.

Some common error situations are described below.

Power-up

No tone on power-up: Check the power adaptor and power source, and restart the phone.

No dial tone on power-up: The MGCP phone takes time to exchange information with DHCP and MGCP call agent. During this time it will play *call progress tone*. Then the tone will change to *dial* if the DHCP and connection with call agent is successful. It will play *error tone* if the DHCP fails or call agent fails.

Plays error tone on power-up: It means that the information exchange with DHCP or MGCP call agent has failed. Check your network connection and confirm that DHCP and MGCP call agent are running. Also, check whether the MGCP phone is configured properly. Restart the phone and check it.

Making a call

The MGCP phone is powered up properly but plays error tone while making call: Check to confirm the network connection and default gateway status.

Plays call-progress tone: MGCP phone will play the call-progress tone while trying to establish a call and this can take time. If it takes a long time, check to make sure the MGCP call agent is running properly.

Plays error tone: The called party may not be registered with the proxy server.

Plays error tone after a long time: Call agent is not running and the MGCP phone times out before playing error tone. This can take some time.

Voice quality is not good: The MGCP phone supports packet loss and network jitter to some extent. Above certain levels, voice quality can deteriorate. The G.729 codec will perform better than the G.711 codec and can be selected in the configuration.

Call hold does not work properly: The call-hold feature requires cooperation from both ends and from the Call Agent. The behavior is not defined if the other phone and Call Agent do not support the hold feature.

Speakerphone does not work: The MGCP phone has a speaker to support one-touch dialing, not for normal speakerphone use. The other party will hear you if you are in speaker mode but the voice on the speaker may be poor quality.

Browser access

No response through browser: Check whether the MGCP phone is connected to the network and you have the correct IP address for the phone.

To access the MGCP phone the IP address is unknown: Select the factory-default option and restart the phone. Now the phone uses factory-default parameters and uses a known IP address.

Browser displays server-busy message: It implies that another person is configuring the MGCP phone.

Browser displays logout message: Check the user name and password. Remember the password is case-sensitive.

Browser displays logout message during configuration: It the browser is idle for more than 10 minutes the MGCP phone will terminate the session. You must restart the browser.

Other functions

Entered factory-default key sequence, no response: You must restart the phone.

Entered production-test key sequence, no response: You must restart the phone. The MGCP phone will exit the production test mode one the next restart.

Entered remote-upgrade key sequence, no response: You need to have the software files in TFTP server and the MGCP phone should be configured with the correct TFTP server and file names.

Entered remote-upgrade key sequence, plays unidentified tone: The MGCP phone plays a tone during software download. The MGCP phone will restart upon successful download.

Power out during remote upgrade: If anything goes wrong during the software upgrade, the phone will use the previous existing software.

MGCP Production Test

This section describes the production test supported by D-Link MGCP phone. The main hardware blocks to be tested are (i) LED, (ii) Key Scan, (iii) Hook Switch, (iv) Codec & Handset, (v) Speaker, (vi) Memory and (vii) Ethernet MAC and PHY. If a test is successful, the MGCP phone will play a *Success* tone and turn on the *Green* LED. If a test fails, it will play an *Error* tone and turn on the *Red* LEDs. After each test, press '1' for going on to the next test and '0' for repeating the test again.
Note: In some tests the MGCP phone cannot determine the outcome of the test and the user must verify it. In such test cases the phone will not play any tone.

LED Test

This is the first test that is performed. This tests the LEDs. In this test, the three LEDs – Mute(*Red*), Hold(*Red*) and Speaker*(Green*) – glow simultaneously for a few seconds and then turn off. No tone is played for this test, as the MGCP phone cannot detect if the test is successful.

Key Scan Test

This tests the keys on the IP Phone. In this test, the user needs to press the keys on the phone in the following order: 0 1 2 3 4 5 6 7 8 9 * # 'mute' 'hold' 'redial' 'speaker'.

Hook Switch Test

This tests the hook switch. In this test, the default case is on-hook. Start the test with 'off-hook' followed by 'on-hook'.

Codec Transmit Test

This test determines if the codec transmission is working properly. In this test, a tone is generated in the handset and speaker simultaneously. It is played till the user doesn't interrupt the test by pressing '1' for going onto the next test or '0' for repeating the same test.

Codec Loop back Test

This test determines if the codec loop back is working properly. In this test, the user must speak into the microphone and that is heard after some finite delay in both the handset and speaker simultaneously. This goes on till the user doesn't interrupt the test by pressing '1' for going onto the next test or '0' for repeating the same test.

SRAM Test

For the SRAM testing, some predefined pattern is written into the data SRAM and program SRAM and is verified after reading from those locations.

Ethernet Transmit Test

In this test, packets containing 1 to 100 as data are transmitted and they are transmitted till the user doesn't interrupt it by pressing any *valid* key on the keypad. This test does not play any tone, since the MGCP phone cannot check if the test is successful.

Ethernet Receive Test

In this test a packet that is sent from the Ethernet driver is received back and is verified. If the test is successful then, it will read *Success*; otherwise an *Error* message appears. It will take some time for *Success* or *Error* to appear, as it takes some time for the driver to receive the packet from network. The user must use a *100 Mbps Switch (full duplex mode)* and connect any two ports for loop-back. *The user must press '0' to repeat the test, and '1' to exit the production test mode.*

Appendix B

DPH-80 SIP (Session Initiation Protocol) IP Phone Configuration

Infrastructure Requirements

Though the DPH-80 SIP phone will work in any type of LAN network, a 100mbps, switched network is more suitable for providing good quality voice communications.

SIP phones need a proxy or redirect server to provide the directory function required to make calls. D-Link SIP phones register the assigned phone number with the server on power up. However, D-Link SIP phones can work through the phone book without an SIP server.

To operate properly, the DPH-80 needs a set of IP parameters such as IP address, subnet mask, gateway address, and DNS server address. These parameters can be configured either statically through a browser or dynamically through DHCP or PPPoE. A DHCP server in the local LAN is required to provide these parameters.

The D-Link SIP phone has many configurable parameters. These parameters can be configured through any Java-enabled Internet browser (Netscape 6.2 or above, IE 5.0 or above).

If your LAN network has a firewall and NAT, they should support SIP to make and receive calls from outside your LAN network.

A TFTP server is required to support remote software upgrading. The TFTP server should have the two software image files (dph80v1.tfp and dph80v2.tfp) from D-Link in the current directory of the TFTP server selected by 'set path'.

Configuring the SIP phone

Once you have the above infrastructure in place, you can power up the SIP phone. The SIP phone will play *call progress tone* and try to register with SIP proxy server. This operation will fail since the SIP phone is not configured properly and it will play *error tone*. However, the SIP phone is accessible through Internet browser for configuration.

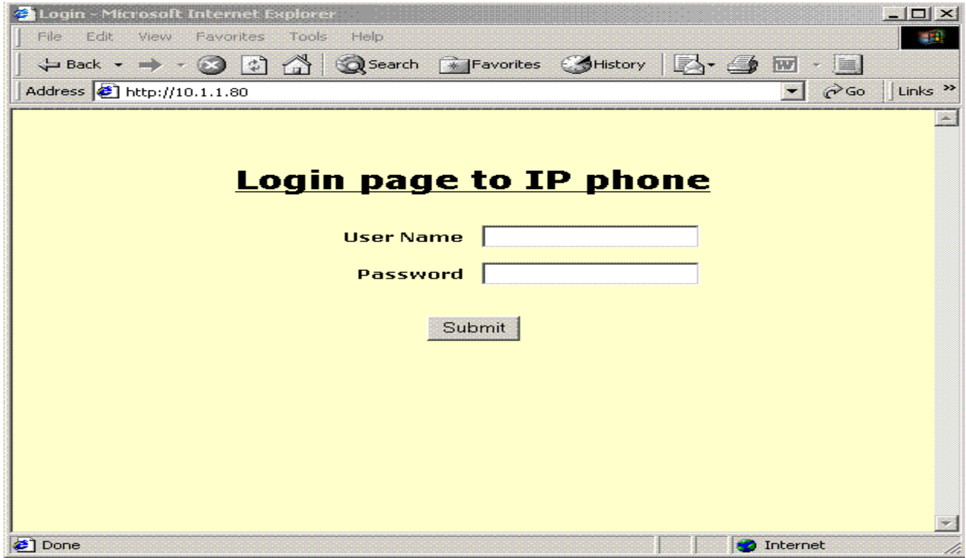
The SIP phone IP address is required to access the phone through a browser. The SIP phone uses factory default values before configuration and the default IP address is 10.1.1.80 (net mask 255.0.0.0). But, the user can enter an IP address through Keypad immediately after factory reset as per the format *x*y*z*a*#, where the symbols * and # are mandatory.

To access the web interface for the D-Link DPH-80:

Use a JavaScript-enabled Internet browser (Netscape 6.2 or above, IE 5.0 or above) with the default IP address of the DPH-80 entered in the address box (*http://10.1.1.80*).

The following page will appear.

Login page to IP phone

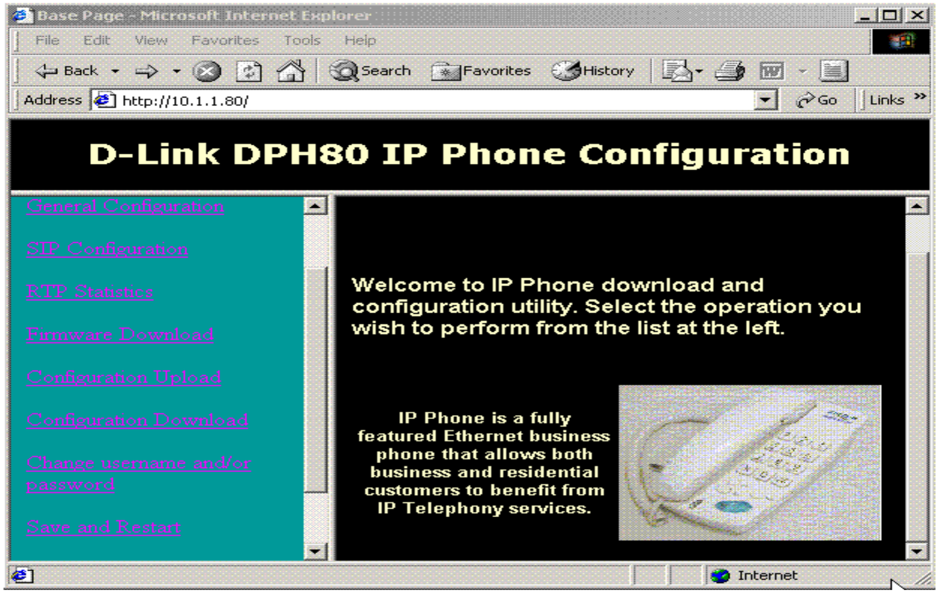


The following two parameters control access to the SIP phone. Default value for both will be "dlink". These values can be changed later using the *Change Login Name and Password* Page.

User Name: This is case-insensitive with a maximum of 20 characters.

Password: This is case-sensitive with a maximum of 20 characters.

After logging in, the *D-Link DPH-80 Phone Configuration* page is displayed and this page provides links to other pages.



Configuration Main Page

Click on General Configuration. A new page containing information about your system and the DPH-80 will appear.

General Configuration

Firmware Version : It shows the current firmware version of the IP phone. It is updated whenever the SIP phone software is updated. It cannot be modified.

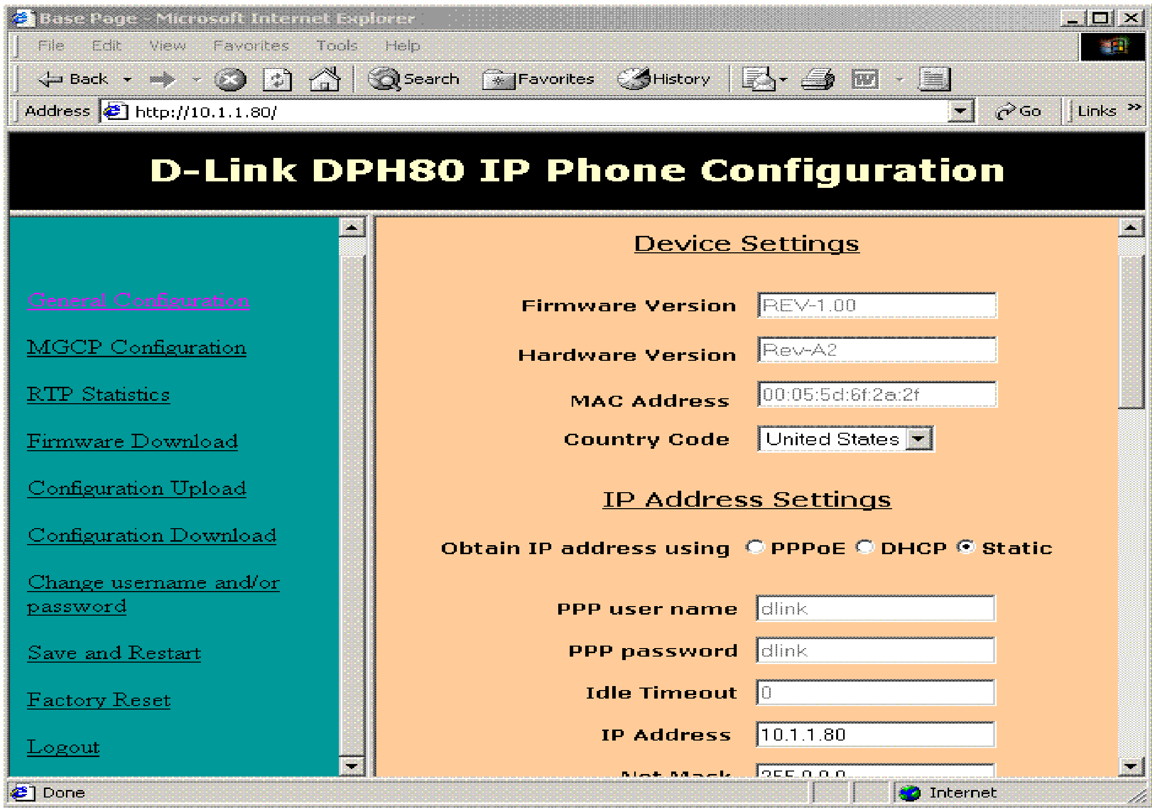
Hardware Version: It shows the current hardware version of the IP phone. It cannot be modified.

MAC Address: This shows the MAC address of the board in colon-separated hex form. Initially it contains the default value ff:ff:ff:ff:ff:ff. It can be modified once to a non-default value. *Once modified, it will be grayed out and cannot be changed again.*

Country Code: This is a drop-down menu. Select the appropriate country. This field controls the type of tones played by the SIP phone.

Obtain IP address using: If *static* option is selected, then a user-configured IP address, Net Mask, Default gateway, and DNS server address would be used for the phone. If *DHCP* is selected, then these values will be obtained using DHCP. If the PPPoE is selected, and using the PPP username and password for authentication, PPPoE obtains an IP address for the phone. Default selection is Static-enabled.

PPP user name: This is the user name used for PPP authentication with the PPP server while obtaining an IP address via PPPoE.



PPP password: This is the password used for PPP authentication with the PPP server while obtaining an IP address via PPPoE.

Idle Timeout: This is the time interval in seconds of session inactivity after which the PPP session should be terminated. If this is set to 0, then the session will never be terminated. This field is grayed out at the moment so that it can't be modified. This will allow the PPP session to be on permanently unless the server closes the connection. This field can be made active later to enable a configuration of timeout value.

IP Address: This should have the IP address of the phone in dot-separated IP address form. An illegal IP address won't be allowed for this field.

Net Mask: This will have the Net Mask of the network to which the IP phone is connected. It must be in dot-separated form. An illegal IP address mask won't be allowed for this field.

Default Gateway: This is the default gateway for the IP phone. An illegal IP address won't be allowed for this field.

DNS server Address: This is the IP address of the DNS server, which will respond to the DNS queries from the IP phone. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

TFTP Server: This has the IP address of the host where TFTP server is running. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

Firmware Filename (up to 6 characters): This is the filename which you want to download from the TFTP server. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Upload Filename (up to 6 characters): This is the filename to upload the configuration parameters from the phone to the TFTP server. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Download Filename (up to 6 characters): This is the filename to download the configuration parameters from the TFTP server to the phone. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Adaptive Jitter: If this is enabled then Jitter Buffer will be adaptive, otherwise it will use a fixed buffer of a size specified in *Maximum Buffer Size.*

Maximum Buffer Size: If adaptive jitter is disabled, the phone will use this static value for Jitter Buffer size. This should be in the range of 0-300 in ms.

Log Server: This flag is turned on in case the user wants to log all debug messages for viewing.

Log Server Address: This has the IP address of the machine where all the log messages should be sent. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

Log Server Port: This is the port number on the log server to which the log messages are to be sent. It should be a valid port number in the range of 0-65335. The user should make sure that it is not one of the reserved port numbers.

Microphone Gain: This will show the microphone gain in the range of -14 to 14 (unit of dB)

Speaker Gain: This will show the speaker gain in the range of -14 to 14 (unit of dB)

Access Settings: The following three key sequences should be unique.

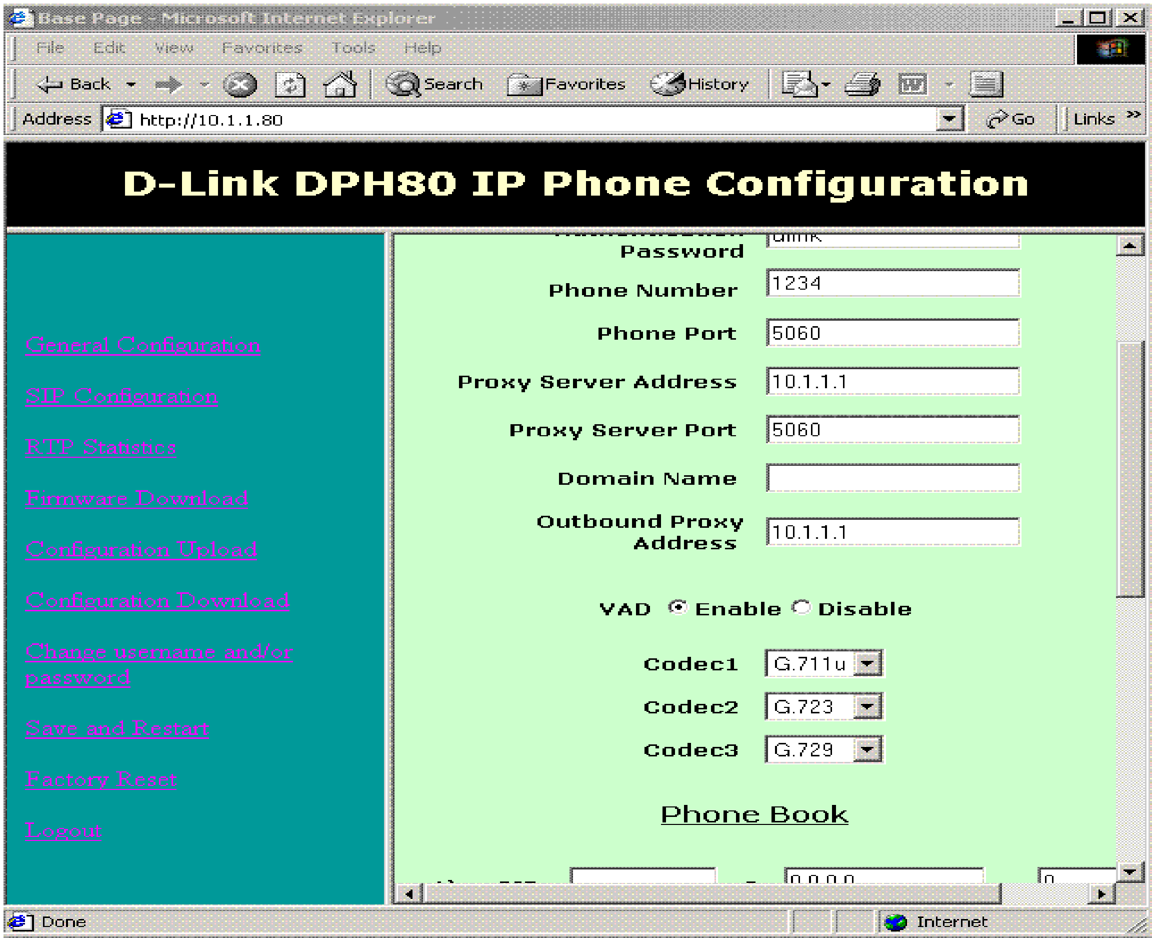
Factory Default: This is the key sequence the user should dial on the phone to get the phone to use all the default values of the parameters. After entering this key sequence on the SIP phone it will restore the parameters to default upon next restart.

Production Key: This is the key sequence the user should dial on the phone to get to production-test mode. After entering this key sequence, the SIP phone will start in production-test mode upon next restart. Reboot after the production test is complete to start functioning in the SIP phone mode.

TFTP upload: This is the key sequence the user should dial on the phone to start the TFTP software update. After getting the new image, the phone will start itself using the new image.

Click on SIP Configuration.

SIP Configuration



User Name: The user name is used to identify the caller for display purposes only. It should be maximum 20 characters and should comprise only letters, digits, hyphen and/or underscore.

Authentication Password: This is used in authentication along with *Phone Number*. It should be maximum 20 characters and should comprise only of letters, digits, hyphen and/or underscore.

Phone Number: This will store any character string up to 20 characters long.

Phone Port: This is the port number at which the phone will open the socket to send and receive SIP messages.

Proxy Server Address: This is the IP address to which all-outgoing SIP messages will be sent. It has to be in dot-separated form. Must be entered for the phone to work with the proxy server and it must be 0.0.0.0 if you want to use phone book without SIP server. It is also used in SIP messages if the *Proxy Domain Name* is null.

Proxy Server Port: This is the port at which the proxy server has opened connection to receive and send packets.

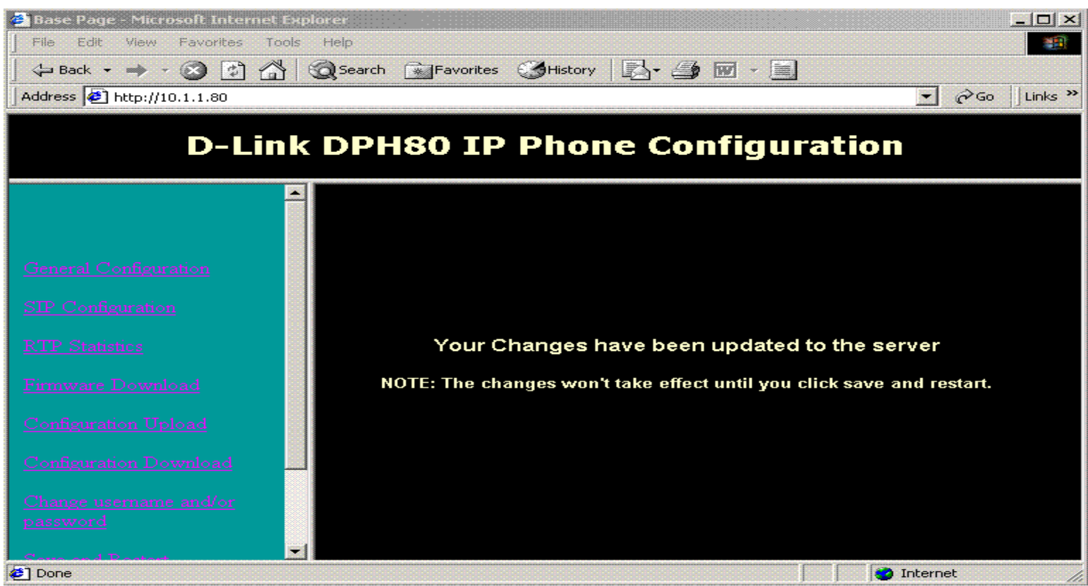
Proxy Domain Name: This is the name of the domain where the IP phone and the proxy/redirect are being hosted. If the field is included it will be used, instead of proxy IP address, in all SIP messages including registration and authentication messages.

Outbound Proxy Address: This is the IP address where SIP messages will be sent. This is useful in traversing a firewall. Normally it should be same as *Proxy Server Address*.
VAD: When this is enabled, the SIP phone uses silence compression to save on bandwidth. This feature works irrespective of the codec selected.

Codec1, Codec2 and Codec3: are drop down menus. This allows selecting what codecs to be used by the phone. It also specifies the priority of the codec while negotiating for the codec to use in any call. Codec1 will be given the highest priority.

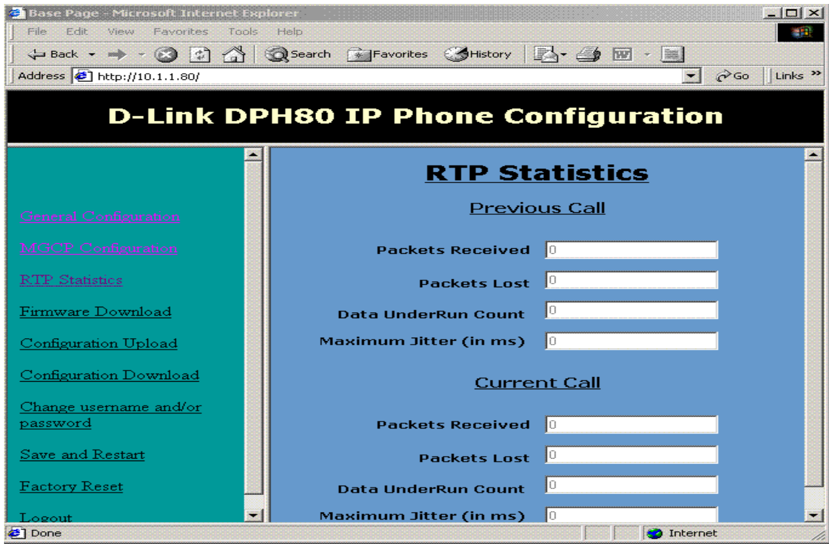
Phone book: This is a table of 10 entries for the phone numbers, IP addresses and ports of the phone. The first field is for the number to be dialed. This can be maximum a 10-digit number. It can contain characters or underscore and hyphen. Next is the IP address of the phone. It should be a valid IP address in dot-separated form. Next is the port number at which the phone is running. It can be any value in the range of 0-65535.

After entering the appropriate values, click Submit. The following page will appear.



Do not click Save and Restart until you have finished configuration.

RTP Statistics



This is an informational page and shows the RTP statistical data from the current call and the previous call. This page is refreshed every 5 seconds automatically.

Packets Received: Number of packets that have been received for the call.

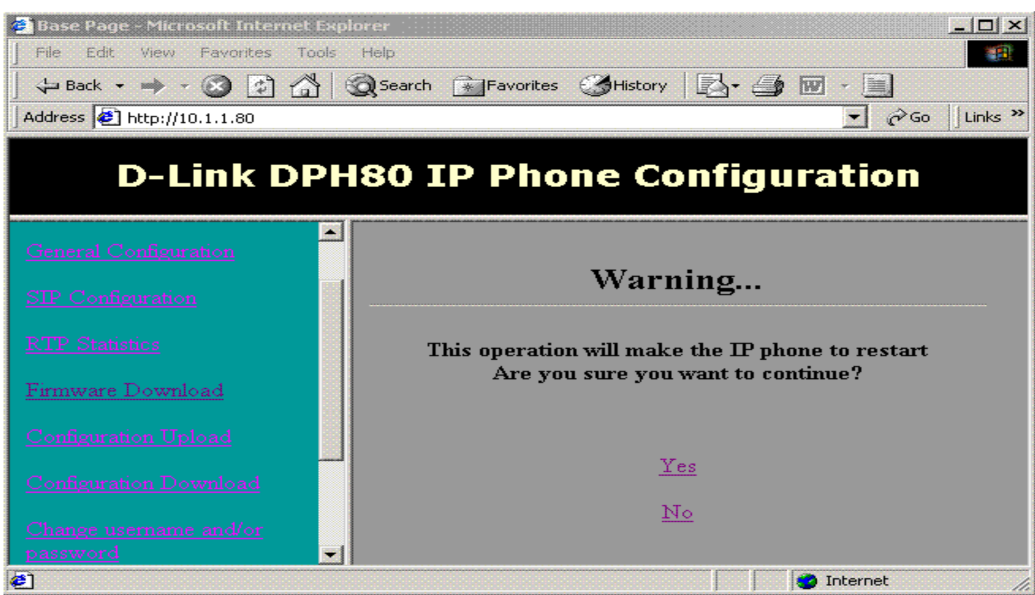
Packets Lost: Number of packets that have been lost in the network.

Data Under Run Count: This is the jitter buffer under run count for the entire call.

Maximum Jitter: This is the estimated maximum jitter in the network, shown in units of ms.

Firmware Download

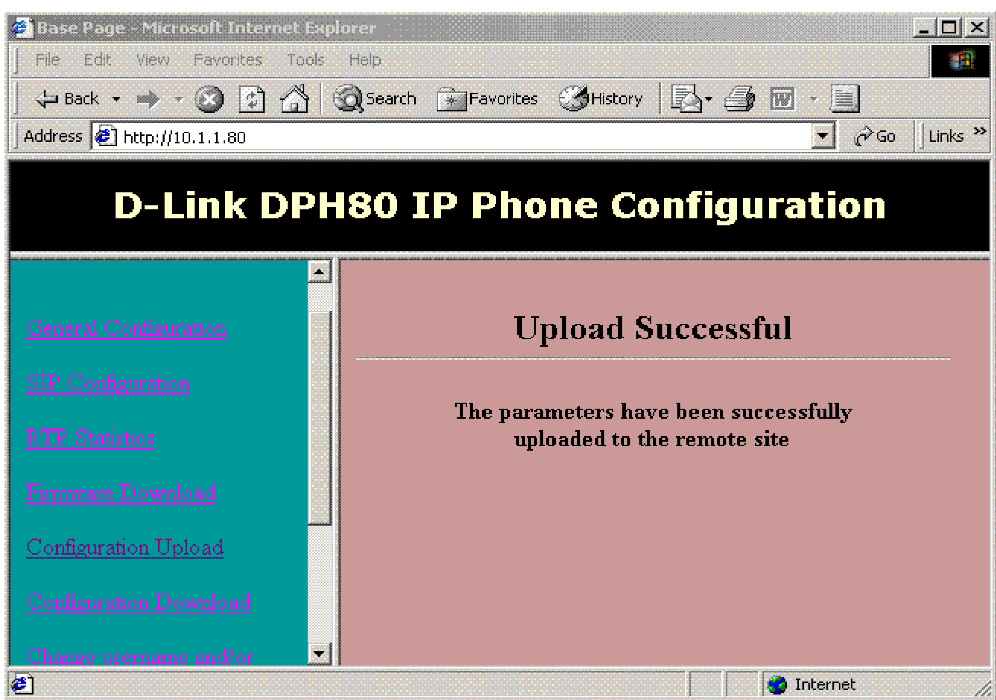
When clicked, this link will display a warning page. Click *'yes'* to download the firmware from the TFTP server to the firmware filename. The TFTP server and filename are set in the General Configuration. Click *'no'* on the warning page to return to the main page.



On clicking Yes the following screen will appear.

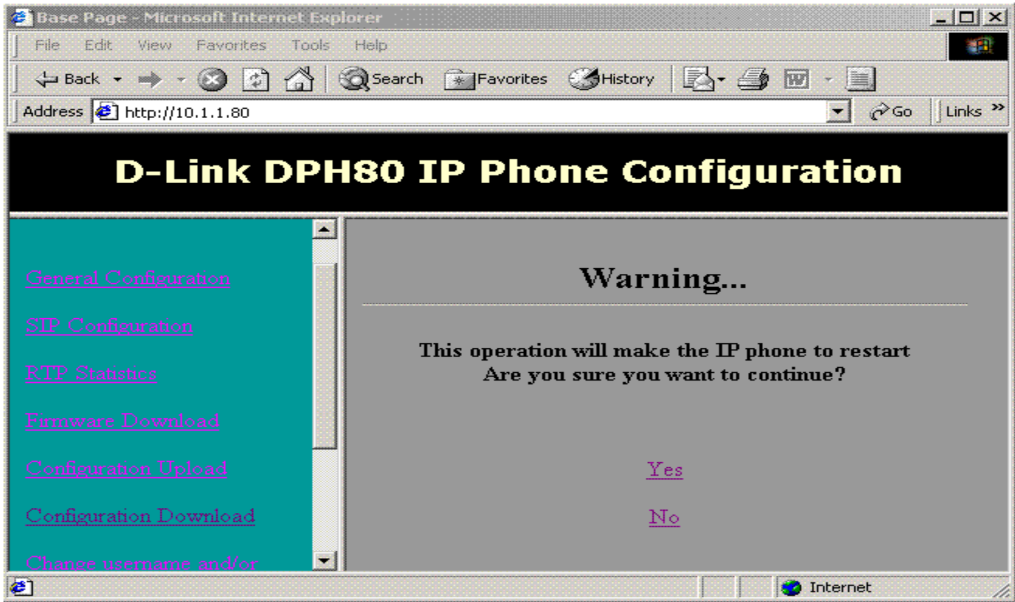
You will be returned to the main page.

Configuration Upload



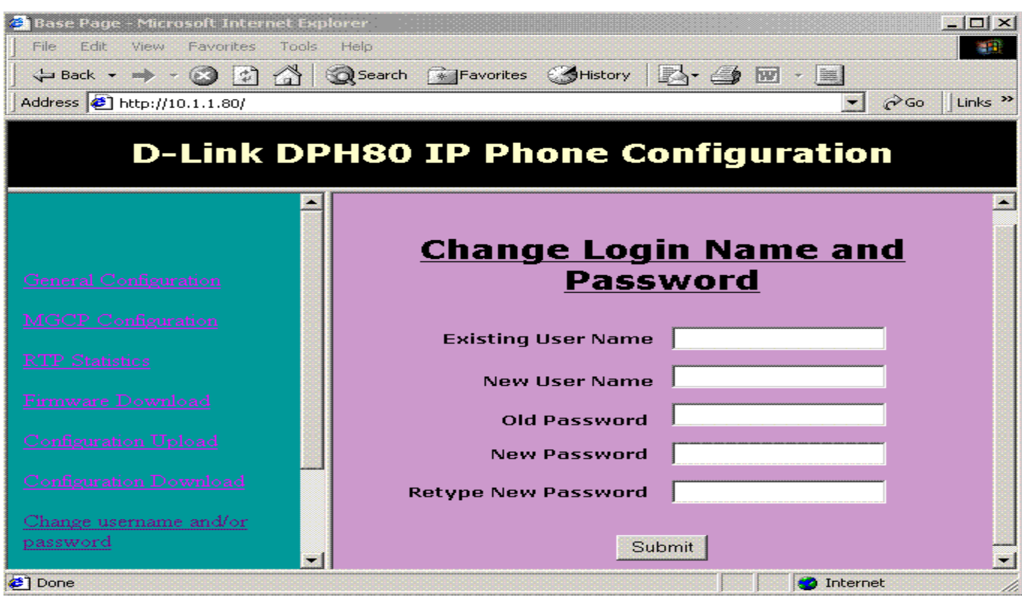
When clicked, this link will display a warning page. Click *'yes'* to upload the configuration parameters from the phone to the TFTP server as the upload filename. The TFTP server and filename are set in the General Configuration. Click '*no'* on the warning page to return to the main page.

Configuration Download



When clicked, this link will display a warning page. Click *'yes'* to download the configuration parameters from the TFTP server to the phone as the download filename. The TFTP server and filename are set in the General Configuration. Click *'no'* on the warning page to return to the main page.

Change Login Name and Password



Existing User Name: This is the user name that was used to access the MGCP phone from the web browser. This is case-insensitive and may be 20 characters long at maximum.

New User Name: If the user wants to change the login user name, it should be entered here. Otherwise, enter the same user name. This is case-insensitive and may be 20 characters long at maximum.

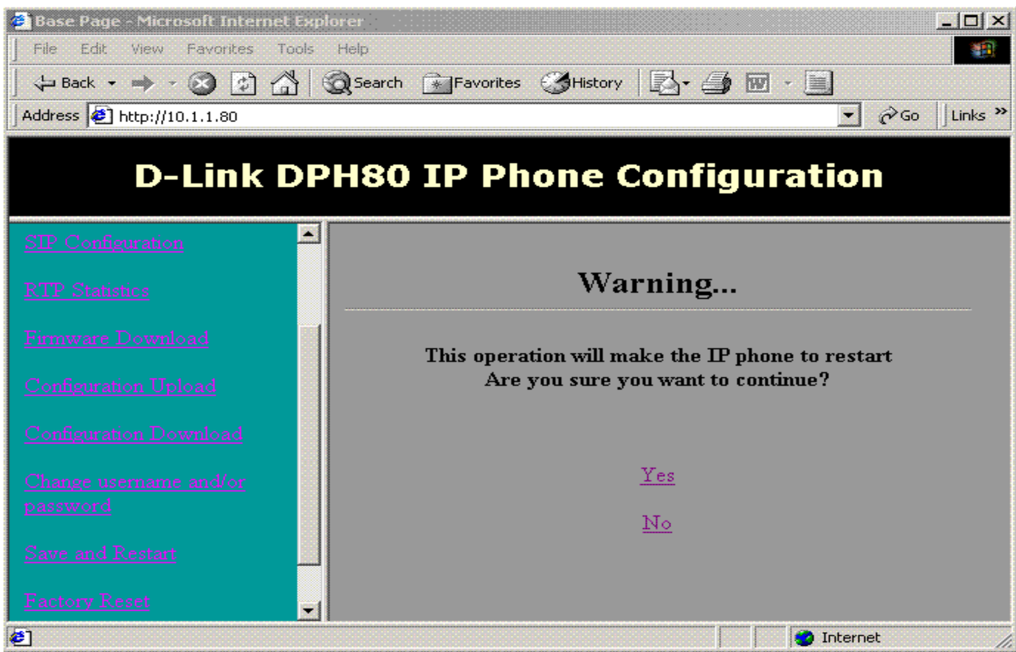
Old Password: This is the login password used to access the MGCP phone from the web browser. This is case-sensitive and may be 20 characters long at maximum.

New Password: A new login password should be entered here. This is case-sensitive and may be 20 characters long at maximum.

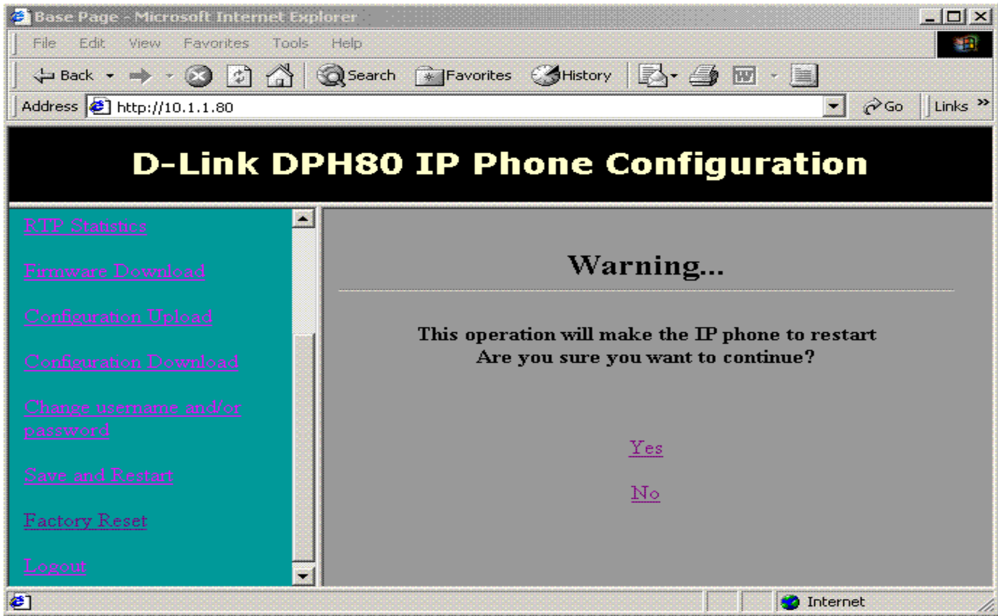
Retype New Password: The above field value should be retyped here to confirm that the correct value was written. If the two don't match, the user will be prompted to retype them.

Click Submit.

Save and Restart

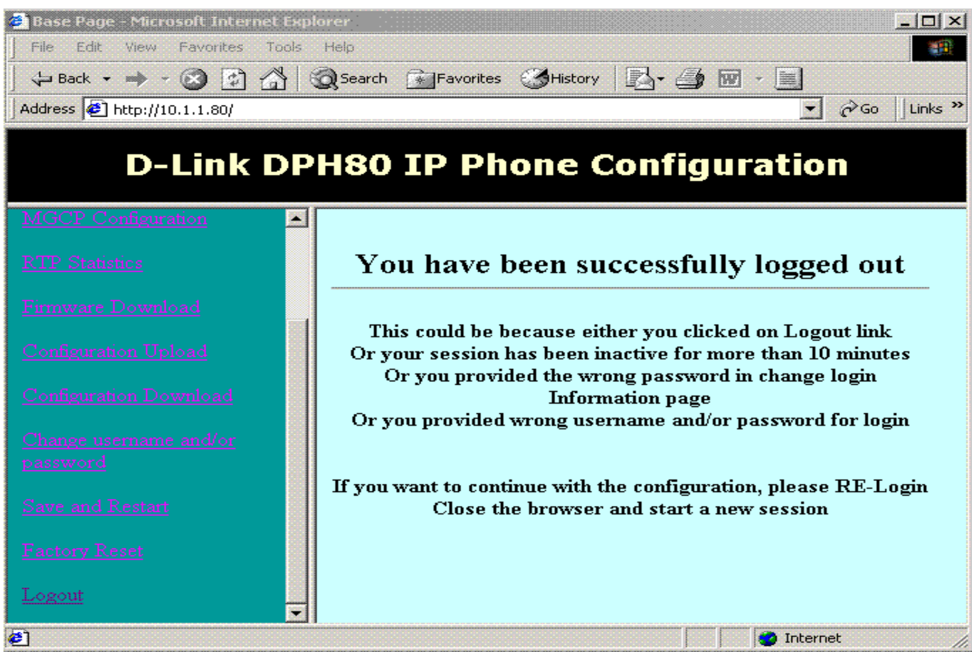


When clicked, this link will display a warning page. Click *'yes'* to save all the updated parameters to the flash memory and restart the phone so that the latest changes take effect. The *You have been successfully logged out* page will be displayed. The phone takes about 30 seconds to come up again. Click *'no'* on the warning page to return to the main page.Factory Reset



When clicked, this link will display a warning page. Click *'yes'* to reset the phone to factory defaults and automatically restart. Click *'no'* on the warning page to return to the main page.

Logout



When clicked, the *You have been successfully logged out* page is displayed and the current configuration session is terminated.

Note: This page may also be displayed if you provided the wrong username and/or password or if your session has been inactive for more than 10 minutes. If you are having an active session with the server, any other user accessing the MGCP phone's configuration will get the *Server Busy* page and will not be allowed access.

Using the SIP Phone

If the SIP phone is configured properly and if the support infrastructure is there, the SIP phone will play dial tone on off-hook. You can dial any registered SIP number by entering the number followed by '#', '#' signals end of dialing to SIP phone.

D-Link SIP phone works in 10, 100, and 10/100 Mbps Ethernet environment. It has an adjustable handset and speaker volume control and it plays tone for all numerical key press.

LEDs

- Link/Activity: Steady on for link up, flashing for activity, off for link down
- Speaker LED (Red): indicates speaker on status
- Hold (Green): Steady on to indicate Hold status; off means normal
- Mute (Red): Steady on to indicate Mute status; off means normal

Tones

D-Link SIP phone plays the following tones depending on phone current state. D-Link SIP supports different type of tones for different country selected through configuration.

- Dial tone
- Call progress tone
- Ring back tone
- Busy tone
- Call alert (ringing) tone
- Error tone
- DTMF tones for all numeric keys

SIP phone Features

- MUTE: When pressed the SIP phone turns off the microphone signal from the handset but still play voice from other party.
- HOLD: When pressed, the SIP phone disconnects both microphone and speaker while connection is kept alive. No voice packets are transmitted from the D-Link SIP phone. Hold LED is on. User may press again to release the call. This feature requires support from the remote phone for proper functioning.
- REDIAL: When pressed, redials last dialed number.
- TRANSFER: Toggle the hook-switch quickly to flash the call. The SIP phone plays dial tone. Then dial the new party's number to transfer the call. The SIP phone transfers the call and plays busy tone. Flashing the hook twice before dialing the number will restore the call to normal state (to call active state).

- SPEAKER: One-touch dialing key. When pressed, speaker LED is on and speaker itself is on while on-hook. If the user off-hooks after dialing or presses the button again, one-touch operation is terminated, LED and speakers turn off.

Algorithms

Codecs: D-Link SIP phones supports G.711 U/A law, G.723.1, and G.729AB. The browser configuration allows selecting the codec and their priority.

Voice activity detection, silence suppression, and comfort noise generation. The VAD can be disabled in the configuration irrespective of the codec being used.

Adjustable Jitter Buffer: D-Link SIP phones uses a robust adaptive jitter buffer algorithm. It can be disabled and a fixed size jitter buffer can be used instead through configuration.

Other features

Remote software upgrade: A predefined key sequence will download the SIP phone software and restart the phone. The SIP phone should have been configured with right TFTP server IP address.

Phone book: This feature allows the phone to be used without a SIP server. A set of 10 numbers can be programmed to the phone and the phone will directly contact these numbers without the help of SIP server.

Remote diagnosis: The SIP phone will send status and other messages to the log server configured in the SIP phone. The remote log server should run the server application from D-Link to receive and display these messages. This feature can be disabled through the browser.

Restore factory defaults: If you enter the specified key sequence, the SIP phone restores the configurable parameters to default values upon next restart.

Production testing: If you enter the specific key sequence, the D-Link SIP phone will execute a production test upon next restart. The production test is described later in this section.

Error Conditions

The D-Link SIP phone detects the following error conditions and plays error tone.

Error tone on network connection failure: It will come to normal state upon network connection. The link LED also gives this information.

Error tone if there is no DHCP server: The phone will recover on detecting a DHCP server.

Error tone if the SIP proxy is down on power up: The phone will recover on detecting a SIP server.

Error tone if the called party is not registered.

SIP Troubleshooting

Some of the common error situations are described below.

Power up

No tone on power up: Check the power adaptor and power source, and restart the phone.

No dial tone on power up: The SIP phone takes time to exchange information with DHCP and SIP proxy servers. During this time it will play *call progress tone*. Then the tone will change to *dial tone* if the exchange is successful or to *error tone* if it is failure. Wait for the tone to change to *dial* or *error tone*.

Plays error tone on power up: It means that the information exchange with DHCP or SIP proxy server has failed. Check if you have a network connection and DHCP and SIP proxy servers are running. Also, check if the SIP phone is configured properly. Restart the phone and check it.

Making call

The SIP phone powered up properly but plays error tone while making call: Check the network connection and default gateway if they are working fine.

Plays call progress tone: SIP phone plays call progress tone while trying to establish call and it can take time. If it takes long time check if the SIP proxy server is running properly.

Plays error tone: The called party may not be registered with the proxy server.

Plays error tone after a long time: The proxy server may not be running and the SIP phone times out before playing error tone. It can take some time.

Voice quality is not good: The SIP phone supports packet loss and network jitter to some extend. Above these levels, the quality can be bad. The G.729 codec will have better performance over the G.711 codec and it can be selected in the configuration.

Call hold does not work properly: The call-hold feature requires co-operation from both ends. The behavior is not defined if the other phone does not support the hold feature.

Speakerphone does not work: The SIP phone has a speaker to support one-touch dialing, not for a speakerphone. The other party will hear you if you are in speaker mode but the voice quality may not be good.

Browser access

No response through browser: Check whether the SIP phone is connected to the network and you have the correct IP address of the phone.

To access the SIP phone if the IP address is unknown: Select the factory-default option and restart the phone. Now the phone will use factory default parameters and uses a known IP address.

Browser displays server-busy message: It implies that another person is configuring the SIP phone.

Browser displays logout message: Check the user name and password. Remember the password is case sensitive.

Browser displays logout message while configuring it: It the browser is idle for more than 10 minutes the SIP phone will terminate the session. You must restart the browser.

Other functions

Entered factory default key sequence, no response: You must restart the phone.

Entered production test key sequence, no response: You must restart the phone. The SIP phone will exit the production test mode next restart.

Entered remote upgrade key sequence, no response: You must have the software files in TFTP server and the SIP phone should be configured with right TFTP server and file names.

Entered remote upgrade key sequence, plays unidentified tone: The SIP phone plays a tone during software download. The SIP phone will restart upon successful download.

Power out during remote upgrade: If anything goes wrong during software upgrade the phone will work the previous existing software.

SIP Production Test

This section describes the production test supported by the D-Link SIP phone. The main hardware blocks to be tested are (i) LED, (ii) Key Scan, (iii) Hook Switch, (iv) Codec & Handset, (v) Speaker, (vi) Memory and (vii) Ethernet MAC and PHY. If a test is successful, the MGCP phone will play a *Success* tone and turn on the *Green* LED. If a test fails, it will play an *Error* tone and turn on the *Red* LEDs. After each test, press '1' for going on to the next test and '0' for repeating the test again.
Note: In some tests the SIP phone cannot determine the outcome of the test and the user must verify it. In such test cases the phone will not play any tone.

LED Test

This is the first test that is performed. This tests the LEDs. In this test, the three LEDs – Mute *(Red)*, Hold *(Red)* and Speaker *(Green)* – glow simultaneously for a few seconds and then turn off. No tone is played for this test, as the SIP phone cannot detect if the test is successful.

Key Scan Test

This tests the keys on the IP Phone. In this test, the user needs to press the keys on the phone in the following order: 0 1 2 3 4 5 6 7 8 9 * # 'mute' 'hold' 'redial' 'speaker'.

Hook Switch Test

This tests the hook switch. In this test, the default case is on-hook. Start the test with 'off-hook' followed by 'on-hook'.

Codec Transmit Test

This test determines if the codec transmission is working properly. In this test, a tone is generated in the handset and speaker simultaneously. It is played till the user doesn't interrupt the test by pressing '1' for going onto the next test or '0' for repeating the same test.

Codec Loop back Test

This test determines if the codec loop back is working properly. In this test, the user must speak into the microphone and that is heard after some finite delay in both the handset and speaker simultaneously. This goes on till the user doesn't interrupt the test by pressing '1' for going onto the next test or '0' for repeating the same test.

SRAM Test

For the SRAM testing, some predefined pattern is written into the data SRAM and program SRAM and is verified after reading from those locations.

Ethernet Transmit Test

In this test, packets containing 1 to 100 as data are transmitted and they are transmitted till the user doesn't interrupt it by pressing any *valid* key on the keypad. This test does not play any tone, since the SIP phone cannot check if the test is successful.

Ethernet Receive Test

In this test a packet that is sent from the Ethernet driver is received back and is verified. If the test is successful, it will read *Success*; otherwise an *Error* message appears. It will take some time for *Success* or *Error* to appear, since it takes time for the driver to receive the packet from network. The user must use a *100 Mbps Switch (full duplex mode)* and connect any two ports for loop-back. *The user must press '0' to repeat the test, and '1' to exit the production test mode.*

Appendix C

DPH-80 H.323 Protocol IP Phone Configuration

Infrastructure Requirements

This section describes the infrastructure requirements for proper functioning of the D-Link (or any) H.323 phone.

- Though the DPH-80 H.323 phones work in any type of LAN network, a 100mbps, switched network is more suitable for providing good quality voice communications.
- H.323 phones need a Gate Keeper (GK) to provide the directory function required to make calls. DPH-80 H.323 phones register the assigned phone number with the server on power-up. However, D-Link H.323 phones can work through the phone book without H.323 GK.
- H.323 phones need a set of IP parameters for proper functioning like an IP address, IP mask, gateway address, and DNS server address. These parameters can be configured either statically through a browser or dynamically through DHCP. A DHCP server in the local LAN is required to provide these parameters.
- The DPH-80 H.323 phone has many configurable parameters. These parameters can be configured through any Java-enabled Internet browser (like Netscape 6.2 or above, IE 5.0 or above).
- If your LAN network has a firewall and NAT, they should support H.323 for making and receiving calls from outside your LAN network.
- A TFTP server is required if you want to support remote software upgrading. The TFTP server should have the two software image files (dph80hi1.tfp and dph80hi2.tfp) from D-Link in the current directory of the TFTP server selected by 'set path'.

Configuring the H.323 phone

Once you have the above infrastructure in place, you can power up the H.323 phone. The H.323 phone will play *call progress tone* and try to register with H.323 Gate Keeper. This operation will fail since the H.323 phone is not configured properly and it will *play error tone*. However, the H.323 phone is accessible through an Internet browser for configuration.

The H.323 phone IP address is required to access the phone through a browser. The H.323 phone uses factory default values before configuration and the default IP address is 10.1.1.80 (net mask 255.0.0.0). The user can enter an IP address immediately after factory reset as per the format *x*y*z*a*#, where the symbols * and # are mandatory.

The following parameters should be configured for proper functioning of the H.323 phone. Other parameters can use default values.

- Phone number (it should be a unique number in H.323 Gate Keeper)
- H.323 Gate Keeper (GK) IP address and port number
- DHCP enable

Save these parameters after modification from the browser. These parameters will be saved to flash and the H.323 phone will restart with new parameters. Now the H.323 phone will play *call progress tone* and try to get IP parameters from DHCP server. If successful, the H.323 phone will try to register with the H.323 Gate Keeper (GK). If the H.323 phone succeeds in the above two operations it will play a dial tone and is ready for use. If either operation fails it will play an error tone and the H.323 phone is not functional.

D-Link H.323 phones can work without H.323 servers through the phone book. For this mode, configure the phone book through browser and the phones will work with configurations in the phone book. Note that phone-book entries of all the phones in one network should be consistent.

D-Link H.323 phones supports a feature whereby you can restart the H.323 phone with factory defaults. This is useful if you want to configure the H.323 phone through the browser and you don't have the H.323 phone IP address. This feature updates the configurable parameters to default values.

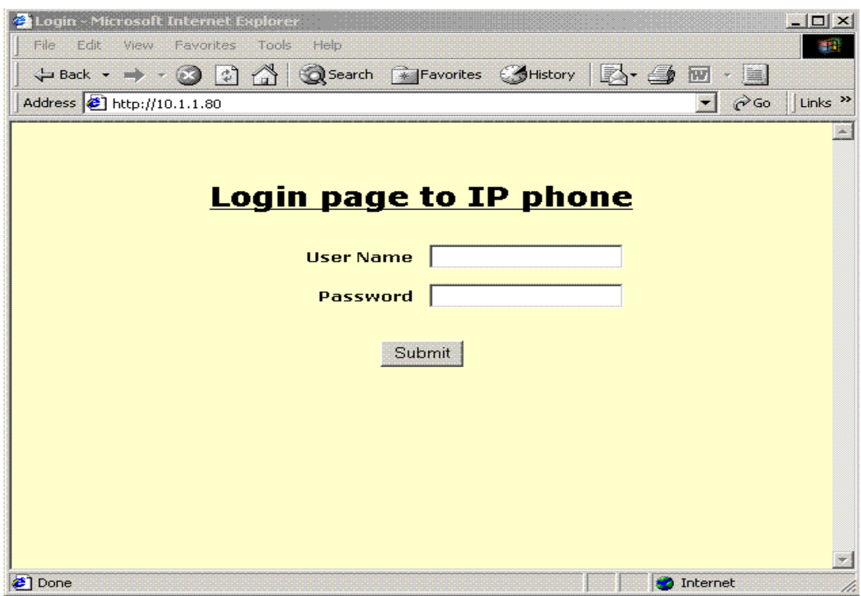
Note: The factory defaults can't be modified after being burned to flash unless the flash is reprogrammed. Thus the factory should burn the correct MAC address to factory defaults. *The software will not work if the MAC chip is other than DL-10022A.*

To access the web interface for the D-Link DPH-80:

Use a JavaScript-enabled Internet browser (Netscape 6.2 or above, IE 5.0 or above) with the default IP address of the DPH-80 entered in the address box (*http://10.1.1.80*).

The following page will appear.

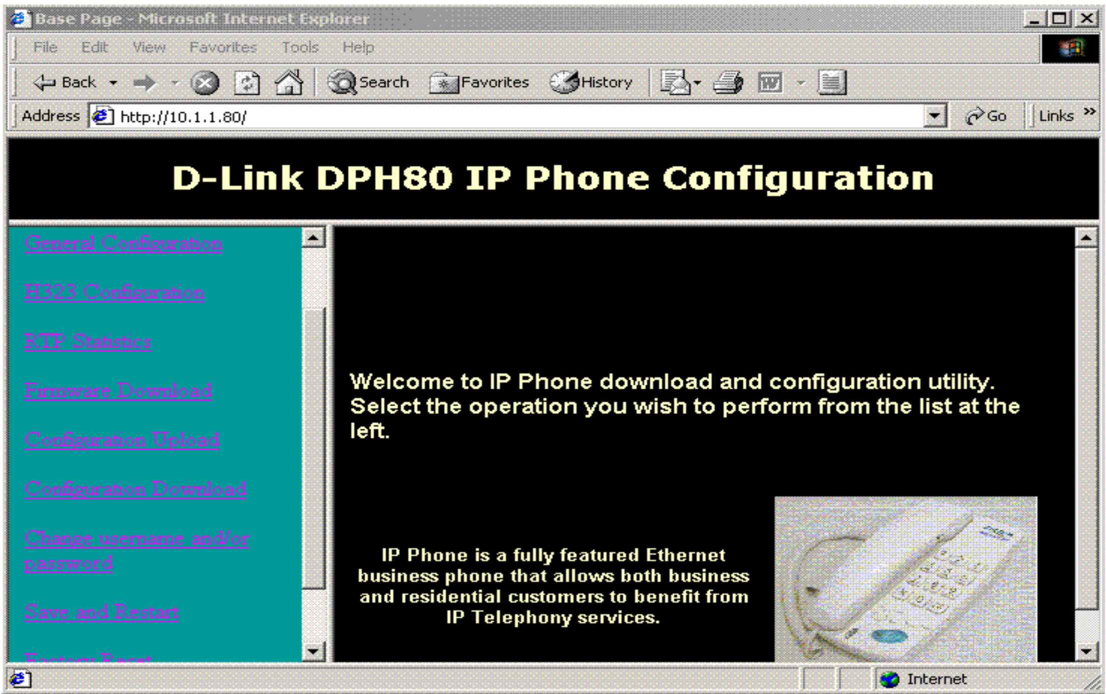
Login Page



The following two parameters control access to the SIP phone. Default value for both will be "dlink". These values can be changed later using the *Change Login Name and Password* Page.

User Name: This is case-insensitive with a maximum of 20 characters.

Password: is case-sensitive with a maximum of 20 characters.



After logging in, the *D-Link DPH-80 Phone Configuration* page is displayed and this page provides links to other pages.

Configuration Main Page

Click on General Configuration. A new page containing information about your system and the DPH-80 will appear.

General Configuration

Firmware Version : It shows the current firmware version of the IP phone. It is updated whenever the SIP phone software is updated. It cannot be modified.

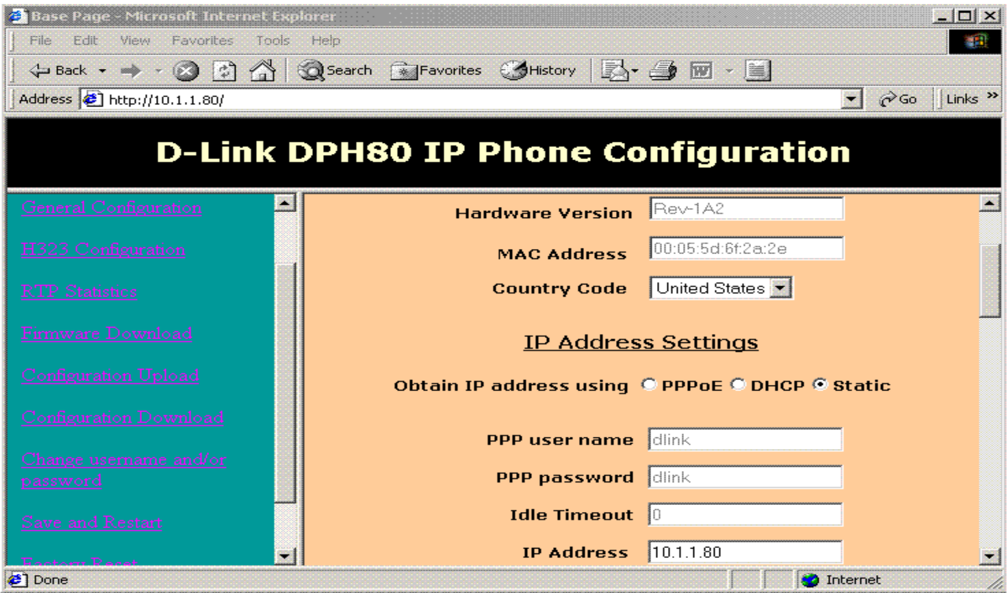
Hardware Version: It shows the current hardware version of the IP phone. It cannot be modified.

MAC Address: This shows the MAC address of the board in colon-separated hex form. Initially it contains the default value ff:ff:ff:ff:ff:ff. It can be modified once to a non-default value. *Once modified, it will be grayed out and cannot be changed again.*

Country Code: This is a drop-down menu. Select the appropriate country. This field controls the type of tones played by the SIP phone.

Obtain IP address using: If *static* option is selected, then a user-configured IP address, Net Mask, Default gateway, and DNS server address would be used for the phone. If *DHCP* is selected, then these values will be obtained using DHCP. If the PPPoE is selected, and using the PPP username and password for authentication, PPPoE obtains an IP address for the phone. Default selection is Static-enabled.

PPP user name: This is the user name used for PPP authentication with the PPP server while obtaining an IP address via PPPoE.



PPP password: This is the password used for PPP authentication with the PPP server while obtaining an IP address via PPPoE.

Idle Timeout: This is the time interval in seconds of session inactivity after which the PPP session should be terminated. If this is set to 0, then the session will never be terminated. This field is grayed out at the moment so that it can't be modified. This will allow the PPP session to be on permanently unless the server closes the connection. This field can be made active later to enable a configuration of timeout value.

IP Address: This should have the IP address of the phone in dot-separated IP address form. An illegal IP address won't be allowed for this field.

Net Mask: This will have the Net Mask of the network to which the IP phone is connected. It must be in dot-separated form. An illegal IP address mask won't be allowed for this field.

Default Gateway: This is the default gateway for the IP phone. An illegal IP address won't be allowed for this field.

DNS server Address: This is the IP address of the DNS server, which will respond to the DNS queries from the IP phone. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

TFTP Server: This has the IP address of the host where TFTP server is running. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

Firmware Filename (up to 6 characters): This is the filename which you want to download from the TFTP server. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Upload Filename (up to 6 characters): This is the filename to upload the configuration parameters from the phone to the TFTP server. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Download Filename (up to 6 characters): This is the filename to download the configuration parameters from the TFTP server to the phone. It may be 6 characters long at maximum. It should start with a letter and should consist of digits, letters and underscore.

Adaptive Jitter: If this is enabled then Jitter Buffer will be adaptive, otherwise it will use a fixed buffer of a size specified in *Maximum Buffer Size.*

Maximum Buffer Size: If adaptive jitter is disabled, the phone will use this static value for Jitter Buffer size. This should be in the range of 0-300 in ms.

Log Server: This flag is turned on in case the user wants to log all debug messages for viewing.

Log Server Address: This has the IP address of the machine where all the log messages should be sent. It must be in dot-separated form. An illegal IP address won't be allowed for this field.

Log Server Port: This is the port number on the log server to which the log messages are to be sent. It should be a valid port number in the range of 0-65335. The user should make sure that it is not one of the reserved port numbers.

Microphone Gain: This will show the microphone gain in the range of -14 to 14 (unit of dB)

Speaker Gain: This will show the speaker gain in the range of -14 to 14 (unit of dB)

Access Settings: The following three key sequences should be unique.

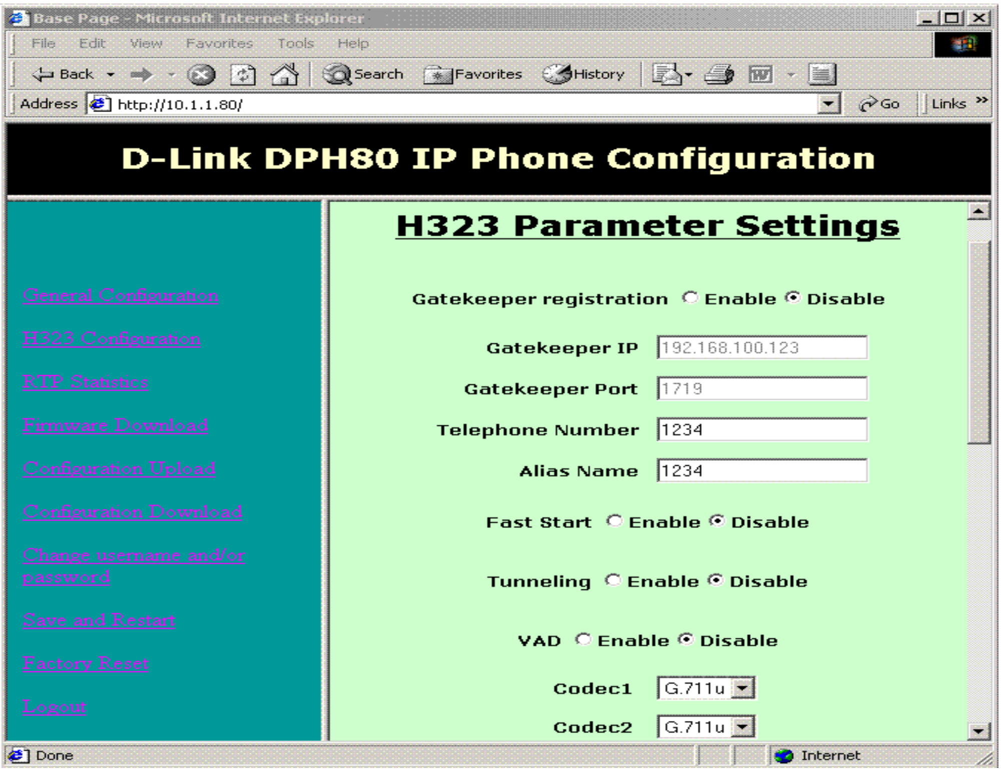
Factory Default: This is the key sequence the user should dial on the phone to get the phone to use all the default values of the parameters. After entering this key sequence on the SIP phone it will restore the parameters to default upon next restart.

Production Key: This is the key sequence the user should dial on the phone to get to production-test mode. After entering this key sequence, the SIP phone will start in production-test mode upon next restart. Reboot after the production test is complete to start functioning in the SIP phone mode.

TFTP upload: This is the key sequence the user should dial on the phone to start the TFTP software update. After getting the new image, the phone will start itself using the new image.

Click on H.323 Configuration.

H.323 Parameters



Gatekeeper registration: Enable and disable option for Gatekeeper registration.

Gatekeeper IP address: IP address of the H.323 Gatekeeper. It must be in dot-separated form. This field is a must for the phone to work with the Gatekeeper (GK).

Gatekeeper Port Number: Port Number of the H.323 Gatekeeper.

Telephone number: Telephone number of this H.323 phone, used in Gatekeeper registration. This will store any alphanumeric character string up to 20 characters.

Alias Name: Alias name for this H.323 phone. This will store any alphanumeric character string up to 30 characters.

Fast Start: Enable and disable option for this mode.

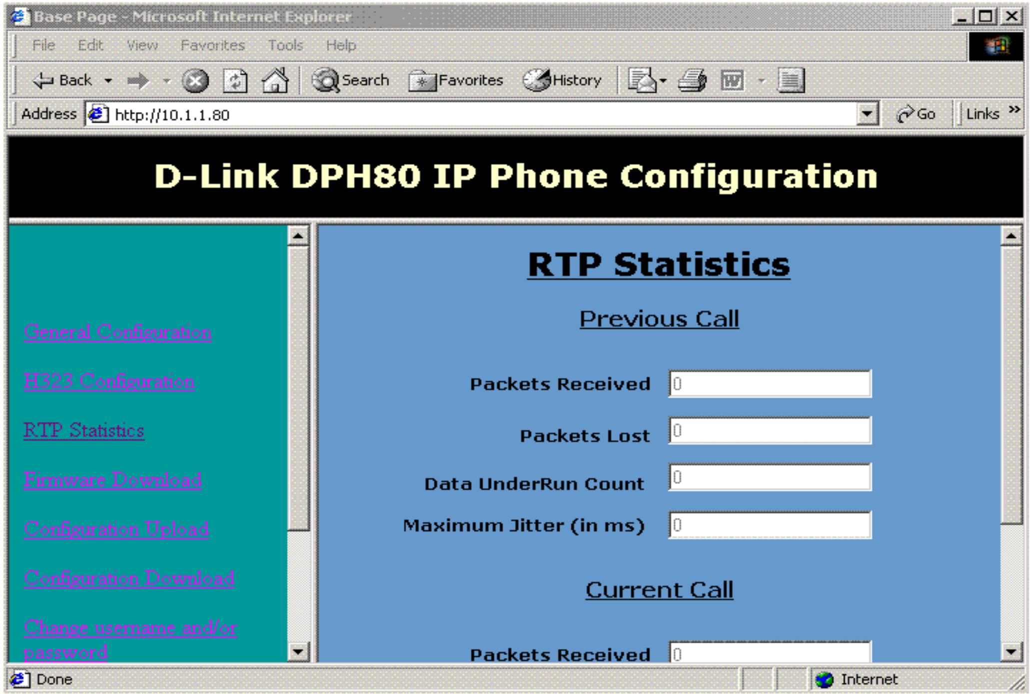
Tunneling: Enable and disable option for this mode.

VAD: When this is enabled, the H.323 phone detects silence interval and uses silence compression to save on bandwidth. This feature works irrespective of the codec selected.

C*odec1, Codec2 and Codec3:* are drop down menus. This allows selecting what codecs to be used by phone. It also specifies the priority of the codec while negotiating for the codec to use in any call. Codec1 will be given the highest priority.

Phone book: This is a table of 10 entries where phone number, IP address and port of the phone may be input. The first field is for the number to be dialed. This can be a maximum 10-digit number. It can contain character or underscore and hyphen. Next is the IP address of the phone we want to dial to. It should be a valid IP address in dot-separated form. Next is the port number at which the phone is running. It can be any value in the range of 0-65535.

RTP Statistics



This is an informational page and shows the RTP statistical data from the current call and the previous call. This page is refreshed every 5 seconds automatically.

Packets Received: Number of packets that have been received for the call.

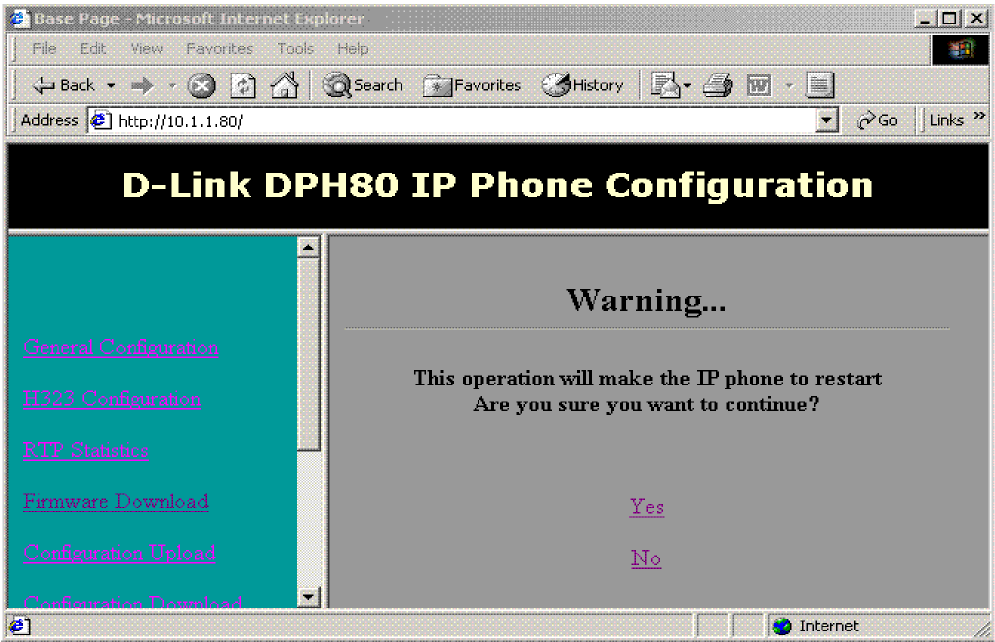
Packets Lost: Number of packets that have been lost in the network.

Data Under Run Count: This is the jitter buffer under run count for the entire call.

Maximum Jitter: This is the estimated maximum jitter in the network, shown in units of ms.

Firmware Download

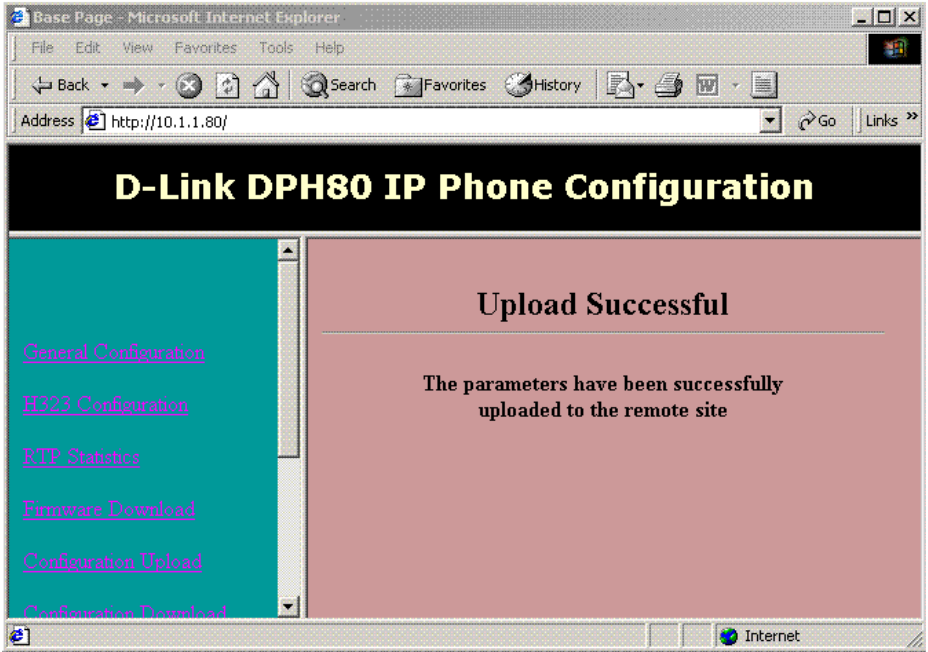
When clicked, this link will display a warning page. Click *'yes'* to download the firmware from the TFTP server to the firmware filename. The TFTP server and filename are set in the General Configuration. Click *'no'* on the warning page to return to the main page.



On clicking Yes the following screen will appear.

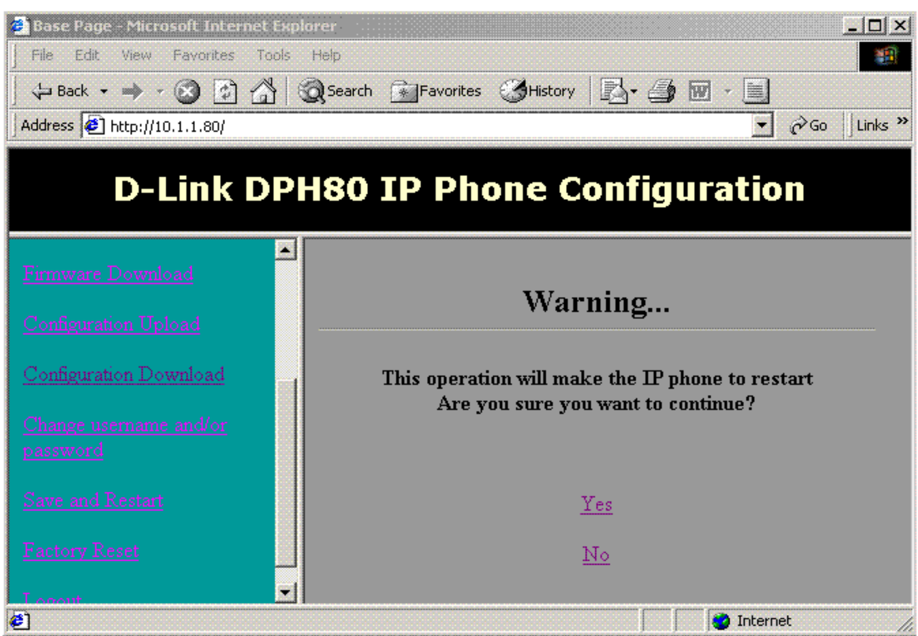
You will be returned to the main page.

Configuration Upload



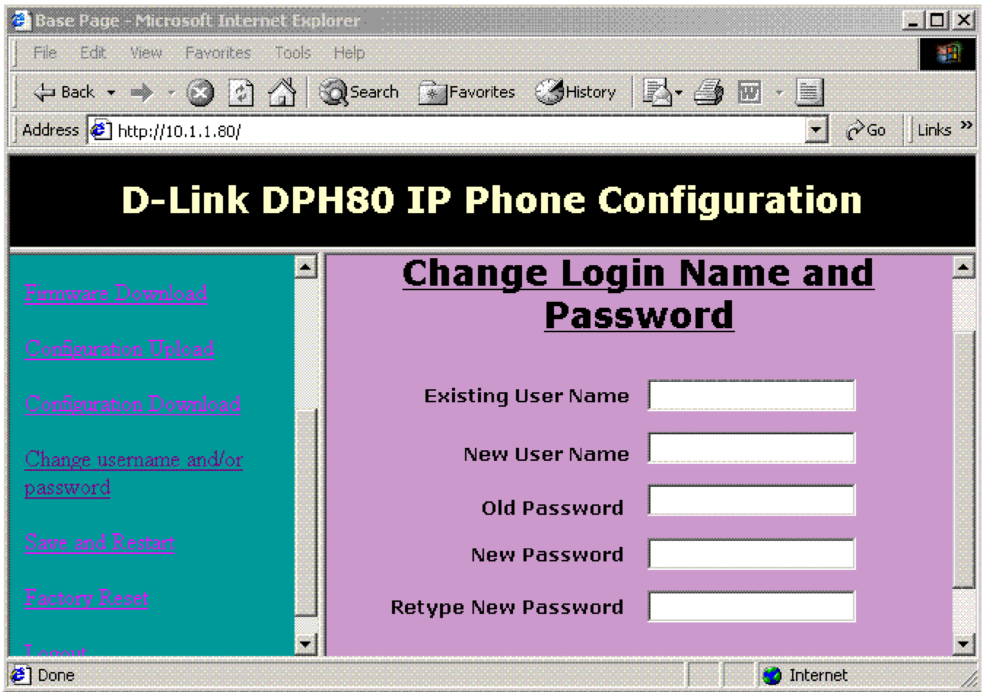
When clicked, this link will display a warning page. Click *'yes'* to upload the configuration parameters from the phone to the TFTP server as the upload filename. The TFTP server and filename are set in the General Configuration. Click '*no'* on the warning page to return to the main page.

Configuration Download



When clicked, this link will display a warning page. Click *'yes'* to download the configuration parameters from the TFTP server to the phone as the download filename. The TFTP server and filename are set in the General Configuration. Click *'no'* on the warning page to return to the main page.

Change Login Name and Password



Existing User Name: This is the user name that was used to access the H.323 phone from the web browser. This is case-insensitive and may be 20 characters long at maximum.

New User Name: If the user wants to change the login user name, it should be entered here. Otherwise, enter the same user name. This is case-insensitive and may be 20 characters long at maximum.

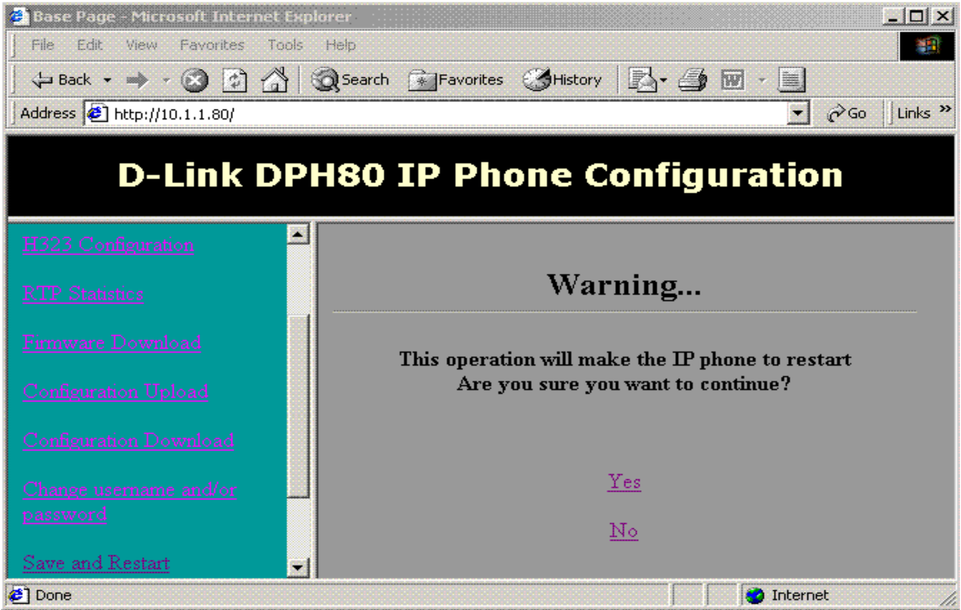
Old Password: This is the login password used to access the DPH-80 phone from the web browser. This is case-sensitive and may be 20 characters long at maximum.

New Password: A new login password should be entered here. This is case-sensitive and may be 20 characters long at maximum.

Retype New Password: The above field value should be retyped here to confirm that the correct value was written. If the two don't match, the user will be prompted to retype them.

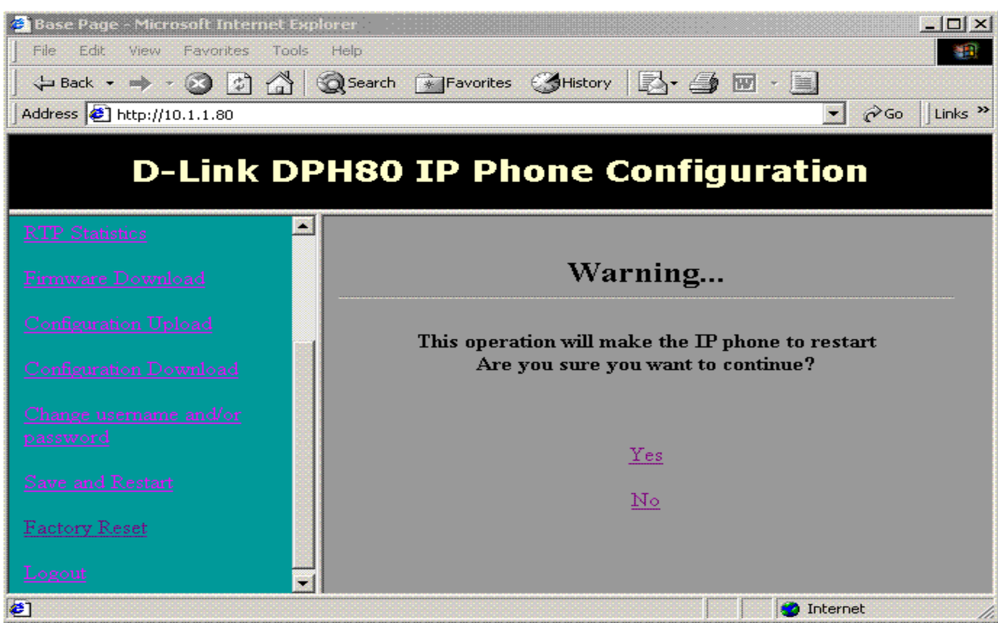
Click Submit.

Save and Restart



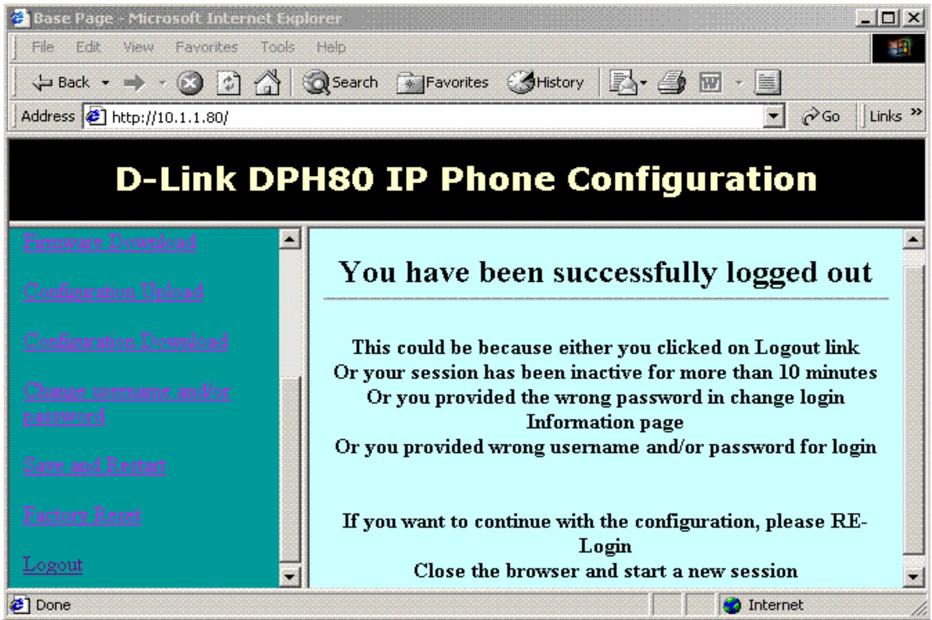
When clicked, this link will display a warning page. Click *'yes'* to save all the updated parameters to the flash memory and restart the phone so that the latest changes take effect. The *You have been successfully logged out* page will be displayed. The phone takes about 30 seconds to come up again. Click *'no'* on the warning page to return to the main page.

Factory Reset



When clicked, this link will display a warning page. Click *'yes'* to reset the phone to factory defaults and automatically restart. Click *'no'* on the warning page to return to the main page.

Logout



When clicked, the *You have been successfully logged out* page is displayed and the current configuration session is terminated.

Note: This page may also be displayed if you provided the wrong username and/or password or if your session has been inactive for more than 10 minutes. If you are having an active session with the server, any other user accessing the H.323 phone's configuration will get the *Server Busy* page and will not be allowed access.

Using the H.323 Phone

If the H.323 phone is configured properly and if the support infrastructure is there, the H.323 phone will play dial tone on off-hook. You can dial any registered H.323 number by entering the number followed by '#' ('#' signals end of dialing to H.323 phone).

D-Link H.323 phone works in 10, 100, and 10/100 Mbps Ethernet environment. It has an adjustable handset and speaker volume control and it plays a tone for all numerical keys.

LEDs

- Link/Activity: Steady on for link up, flashing for activity, off for link down
- Speaker LED (Red): indicates speaker on status
- Hold (Green): Steady on to indicate Hold status; off means normal
- Mute (Red): Steady on to indicate Mute status; off means normal

Tones

D-Link H.323 phone plays the following tones depending on the phone's current state. The DPH-80 supports different type of tones for different countries – selected through configuration.

- Dial tone
- Call progress tone
- Ring back tone
- Busy tone
- Call alert (ringing) tone
- Error tone
- DTMF tones for all numeric keys

Phone Features

- MUTE: When pressed the H.323 phone turns off the microphone signal from the handset but still play voice from other party.
- HOLD: When pressed, the H.323 phone disconnects both microphone and speaker while connection is kept alive. No voice packets are transmitted from the D-Link DPH-80 phone. Hold LED is on. User may press again to release the call. This feature requires support from the remote phone for proper functioning.
- REDIAL: When pressed, redials last dialed number.
- TRANSFER: Toggle the hook-switch quickly to flash the call. The phone plays the dial tone. Then dial the new party's number to transfer the call. The H.323 phone transfers the call and plays busy tone. Flashing the hook twice before dialing the number will restore the call to normal state (to call-active state).
- SPEAKER: One-touch dialing key. When pressed, speaker LED is on and speaker itself is on while on-hook. If the user off-hooks after dialing or presses the button again, one-touch operation is terminated, LED and speakers turn off.

Algorithms

Codecs: D-Link H.323 phones supports G.711 U/A law, G.723.1, and G.729AB. The browser configuration allows selecting the codec and their priority.

Voice activity detection, silence suppression, and comfort noise generation. The VAD can be disabled in the configuration irrespective of the codec being used.

Adjustable Jitter Buffer: D-Link H.323 phones uses a robust adaptive jitter buffer algorithm. It can be disabled and a fixed size jitter buffer can be used instead through configuration.

Other features

Remote software upgrade: A predefined key sequence will download the H.323 phone software and restart the phone. The DPH-80 should have been configured with right TFTP server IP address.

Phone book: This feature allows the phone to be used without a H.323 server. A set of 10 numbers can be programmed to the phone and the phone will directly contact these numbers without the help of H.323 server.

Remote diagnosis: The H.323 will send status and other messages to the log server configured in the DPH-80. The remote log server should run the server application from D-Link to receive and display these messages. This feature can be disabled through the browser.

Restore factory defaults: If you enter the specified key sequence, the H.323 phone restores the configurable parameters to default values upon next restart.

Production testing: If you enter the specific key sequence, the D-Link H.323 phone will execute a production test upon next restart. The production test is described later in this section.

Error Conditions

The D-Link H.323 phone detects the following error conditions and plays the error tone.

Error tone on network connection failure: It will return to normal state upon network connection. The link LED also gives this information.

Error tone if there is no DHCP server: The phone will recover on detecting a DHCP server.

Error tone if the SIP proxy is down on power up: The phone will recover on detecting a SIP server.

Error tone if the called party is not registered.

H.323 Troubleshooting

Some of the common error situations are described below.

Power up

No tone on power up: Check the power adaptor and power source, and restart the phone.

No dial tone on power up: The H.323 phone takes time to exchange information with DHCP and H.323 Gatekeeper (GK). During this time it will play *call progress tone*. Then the tone will change to *dial tone* if the exchange is successful or to *error tone* if it is failure. Wait for the tone to change to *dial or error tone*.

Plays error tone on power up: It means that the information exchange with DHCP or H.323 Gatekeeper (GK) has failed. Check if you have network connection and DHCP and H.323 Gatekeeper (GK) are running. Also, check whether the H.323 phone is configured properly. Restart the phone and check it.

Making call

The H.323 phone is powered up properly but plays error tone while making call: Check the network connection and default gateway.

Plays call progress tone: H.323 phone plays call progress tone while trying to establish call and it can take time. If it takes long time check if the H.323 Gate Keeper (GK) is running properly.

Plays error tone: The called party may not be registered with the Gate Keeper (GK).

Plays error tone after long time: The Gate Keeper (GK) may not be running and the H.323 phone times out before playing error tone. It can take some time.

Voice quality is not good: The H.323 phone supports packet loss and network jitter to some extend. Above these levels, the quality can be bad. G.729 codec will have better performance over G.711 codec and may be selected in the configuration.

Call hold does not work properly: Call-hold feature require cooperation from both ends. The behavior is not defined if the other phone does not support the hold feature.

Speakerphone does not work: The H.323 phone has a speaker to support one-touchdialing, not for speakerphone. The other party will hear you if you are in speaker mode and voice on speaker may not be good.

Browser access

No response through browser: Make sure the H.323 phone is connected to the network and you have the correct IP address of the phone.

To access the H.323 phone if the IP address is unknown: Select the factory default option and restart the phone. Now the phone uses factory default parameters and it uses a known IP address.

Browser displays server-busy message: It implies that another person is configuring the H.323 phone.

Browser displays logout message: Check the user name and password. Remember the password is case sensitive.

Browser displays logout message while configuring it: It the browser is idle for more than 10 minutes the H.323 phone will terminate the session. You must restart the browser.

Other functions

Entered factory default key sequence, no response: You must restart the phone.

Entered production test key sequence, no response: You must restart the phone. The H.323 phone will exit the production test mode next restart.

Entered remote upgrade key sequence, no response: You need to have the software files in TFTP server and the H.323 phone should be configured with the correct TFTP server and file names.

Entered remote upgrade key sequence, plays unidentified tone: The H.323 phone plays a tone during software download. The H.323 phone will restart upon successful download.

Power out during remote upgrade: If anything goes wrong during software upgrade the phone will use the previous existing software.

H.323 Production Test

This section describes the production test supported by the D-Link H.323 phone. The main hardware blocks to be tested are (i) LED, (ii) Key Scan, (iii) Hook Switch, (iv) Codec & Handset, (v) Speaker, (vi) Memory and (vii) Ethernet MAC and PHY. If a test is successful, the MGCP phone will play a *Success* tone and turn on the *Green* LED. If a test fails, it will play an *Error* tone and turn on the *Red* LEDs. After each test, press '1' for going on to the next test and '0' for repeating the test again.

Note: In some tests the H.323 phone cannot determine the outcome of the test and the user must verify it. In such test cases the phone will not play any tone.

LED Test

This is the first test that is performed. This tests the LEDs. In this test, the three LEDs – Mute *(Red)*, Hold *(Red)* and Speaker *(Green*) – glow simultaneously for a few seconds and then turn off. No tone is played for this test, as the H.323 phone cannot detect if the test is successful.

Key Scan Test

This tests the keys on the IP Phone. In this test, the user needs to press the keys on the phone in the following order: 0 1 2 3 4 5 6 7 8 9 * # 'mute' 'hold' 'redial' 'speaker'.

Hook Switch Test

This tests the hook switch. In this test, the default case is on-hook. Start the test with 'off-hook' followed by 'on-hook'.

Codec Transmit Test

This test determines if the codec transmission is working properly. In this test, a tone is generated in the handset and speaker simultaneously. It is played till the user doesn't interrupt the test by pressing '1' for going onto the next test or '0' for repeating the same test.

Codec Loop back Test

This test determines if the codec loop back is working properly. In this test, the user must speak into the microphone and that is heard after some finite delay in both the handset and speaker simultaneously. This goes on till the user doesn't interrupt the test by pressing '1' for going onto the next test or '0' for repeating the same test.

SRAM Test

For the SRAM testing, some predefined pattern is written into the data SRAM and program SRAM and is verified after reading from those locations.

Ethernet Transmit Test

In this test, packets containing 1 to 100 as data are transmitted and they are transmitted till the user doesn't interrupt it by pressing any *valid* key on the keypad. This test does not play any tone, since the MGCP phone cannot check if the test is successful.

Ethernet Receive Test

In this test a packet that is sent from the Ethernet driver is received back and is verified. If the test is successful then, it will read *Success*; otherwise an *Error* message appears. It will take some time for *Success* or *Error* to appear, as it takes some time for the driver to receive the packet from network. The user must use a *100 Mbps Switch (full duplex mode)* and connect any two ports for loop-back. *The user must press '0' to repeat the test, and '1' to exit the production test mode.*