Product photo for illustration only. Actual appearance may vary by hardware version and regional release.

User Manual*

1-Line PoE SIP IP Phone

* Features, menu names, defaults, illustrations and accessories may vary with hardware revision, firmware, region and service-provider configuration. Use the settings available on your device and the documentation for its exact revision. Illustrations are examples and do not establish the number of accounts, ports or programmable keys. GUI views may be cropped to show only the controls discussed. Provider-dependent features require a compatible SIP service. Optional accessories are supplied separately unless your package states otherwise. For product support and revision-specific downloads, visit dlink-mea.com.

For support, visit dlink-mea.com

DPH-115SE / DPH-115GE

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1. Safety and installation precautions

Read these instructions before connecting the phone. Keep the phone indoors on a stable, dry, ventilated surface, away from direct sunlight, heaters and liquids.

- Use a D-Link-approved power adapter that matches the voltage, current and polarity marked on the phone. Check the mains input rating on the adapter before connecting it.
- Connect only the low-voltage DC output of the adapter to the phone. Never connect mains AC power directly to the phone.
- Do not twist, stretch, crush or damage power and network cables. Stop using a damaged cable or plug.
- Keep the power plug accessible so that the equipment can be disconnected quickly. Disconnect power before cleaning or moving the phone.
- If the phone has been moved from a cold environment to a warm one, allow any condensation to dry completely before applying power.
- Do not cover ventilation openings or place the phone on cushions or carpets. Keep packaging bags away from children.
- Do not insert objects into openings or attempt repairs. Contact an authorized D-Link service provider if the unit is damaged or a liquid or foreign object enters it.
- During a thunderstorm, avoid touching exposed connectors, the power plug and connected telephone or network lines.
- Clean the disconnected phone with a soft cloth slightly dampened with water. Do not use solvents or abrasive cleaners.
- For limits such as operating temperature and humidity, follow the label and the datasheet for the exact hardware revision. Do not operate outside those limits.

2. Getting started

The D-Link DPH-115 is an office IP phone with a graphical LCD, handset and hands-free calling. It supports SIP calling, call history, phonebooks, transfer, hold, forwarding and conferencing through a compatible IP telephony service.*

* Programmable-key count, display/backlight options, account capacity and optional ports depend on hardware and firmware. This manual does not treat a key slot, a soft key and a simultaneous call as the same specification.

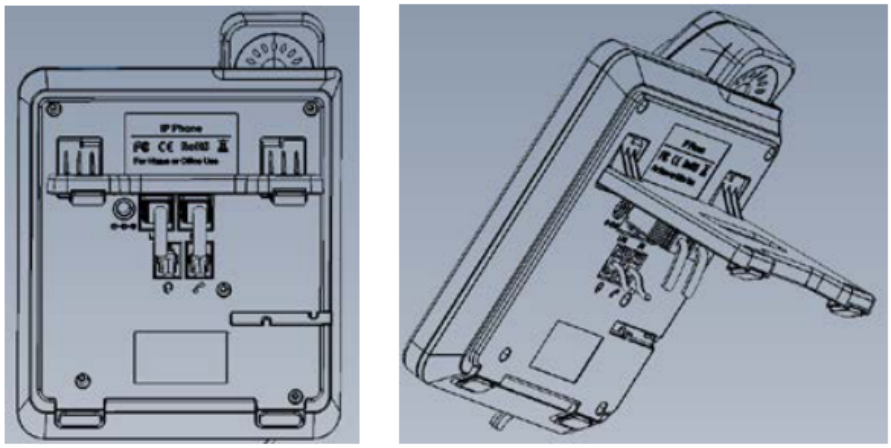
Package contents

Check for the DPH-115 phone, handset, handset cord, Ethernet cable, stand and installation information supplied with your package. Contact your supplier if a listed included item is missing. A power adapter or other optional accessory may be supplied separately.

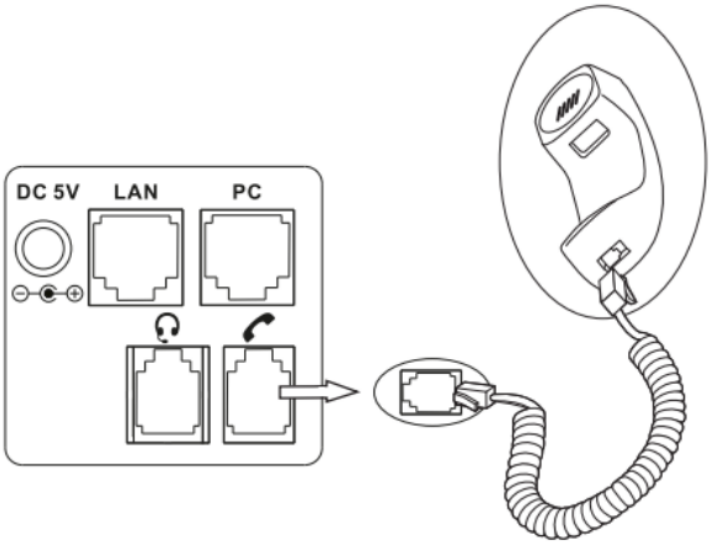
Fit the stand and connect cables

1. Disconnect power. Align the stand with the mounting points on the rear of the phone and secure it at the supported viewing angle.

2. Connect the handset cord to the handset and the handset socket. Connect an optional supported headset only to a headset-labeled socket.
3. Connect the network cable to the network-facing LAN port shown in the illustration. A computer may connect to the PC port.
4. For a PoE-capable revision, use a compatible IEEE 802.3af power source. Otherwise connect the approved DC adapter. Check the product label before choosing the power method.



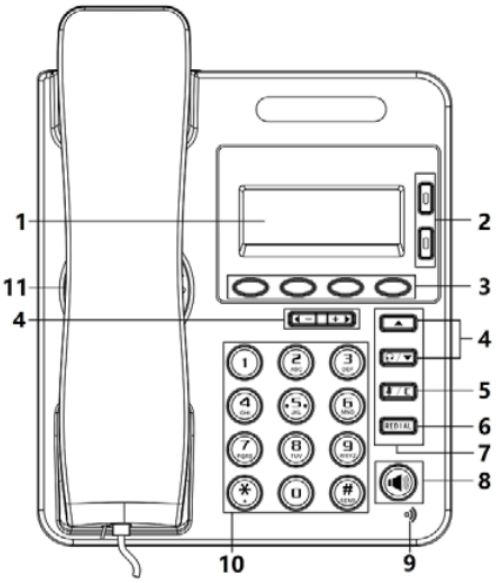
*Stand-fitting illustration**



*Connection illustration - use the labels on your actual device**

* The DPH-115 is an IP phone. Do not connect a PSTN line to an Ethernet or handset connector. Optional sockets or power capabilities must be checked on the device label and matching datasheet.

3. Controls and screen



*Front-panel illustration; numbered controls are described below**

Setting	Description
1. LCD	Shows account status, call information, soft-key labels, time and notifications.
2. Line keys	Selects a line/account or an assigned function. LED indications depend on call and registration state.
3. Soft keys	Performs the action shown immediately above the key on the screen.
4. Navigation / Volume	Moves through menus and lists. Direction-key volume behavior depends on the current screen.
5. Mute / C	Cancels or returns from an operation. During a call, the mute control stops microphone transmission. In idle mode, C can show phone status on the described firmware.
6. Redial	Redials a previous number or opens a recent-call action.
7. Programmable controls	Provides configurable shortcuts where the firmware allows a long press or a web key assignment.
8. Speaker	Activates hands-free mode. On the described firmware, a long press selects headset mode where supported.
9-11. Microphone, keypad, speaker	Provides voice input, number/character entry and hands-free audio.

Status icons indicate registration, DND, missed calls, forwarding, messages, mute, audio mode, keypad lock and network availability. Read the status text or icon together with the current screen; indicator colors alone

may not distinguish every condition.

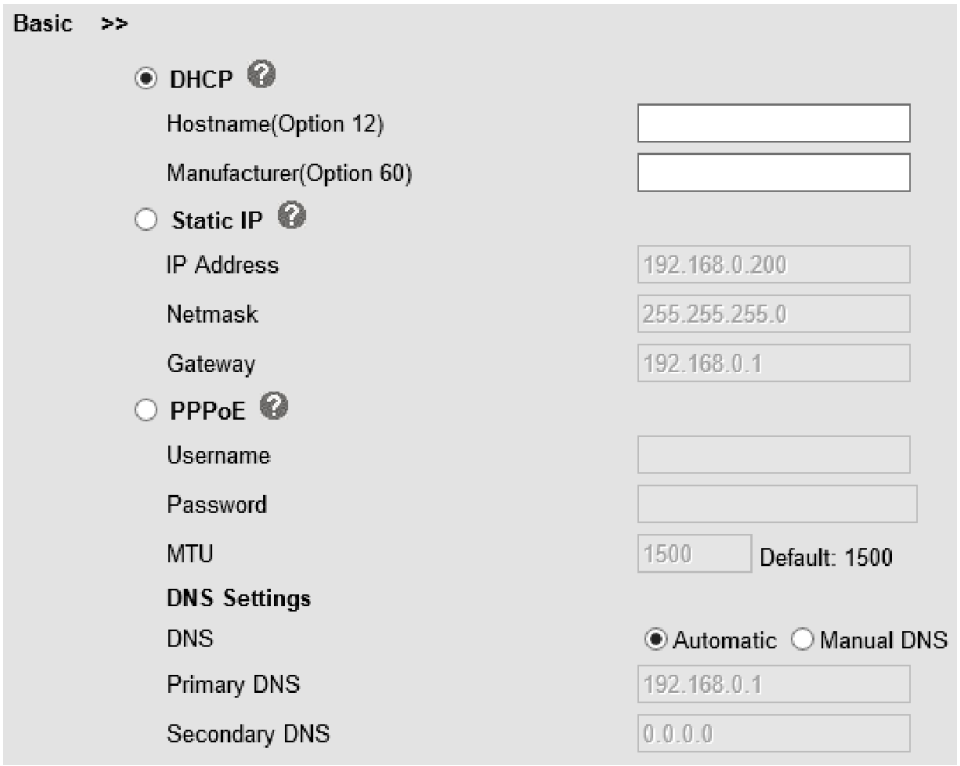
4. Network and first login

Choose a network configuration

Setting	Description
DHCP	Recommended when a DHCP service is available. The network supplies the IP address, subnet mask, gateway and DNS settings.
Static IP	Enter the IP address, subnet mask, default gateway and DNS servers supplied by your network administrator. Use an address that does not conflict with another device.
PPPoE	Use only when required by the service provider. Enter the provider-issued username and password.

Connect the network switch or router to the phone's network-facing LAN port. The PC PC-facing port, where fitted, connects a computer through the phone. Follow the labels on your device; do not connect the uplink to the PC-facing port by mistake.

* Port labels, link speeds, PoE availability and PC-port behavior vary by revision. Check the matching device label and datasheet. After changing the IP address, reconnect the browser using the new address. Network changes may require a reboot.



Network connection and DNS settings (cropped). Example values only; use the settings assigned to your device and service.*

Configure through the phone

1. Press Menu and open System Settings > Advanced Settings > Network > LAN Port. Enter the current administrator/menu password if requested. The documented factory menu password is empty; provisioned phones may differ.
2. Choose DHCP, Static or PPPoE. Enter only the settings required for that mode. Use * in place of a dot only where the keypad IP-entry field requires it.
3. Save the changes. If the phone requests a restart, confirm it and wait for startup to finish.
4. In idle mode, press C to display phone status and note the assigned IP address.

Open the web interface

Connect a computer to the same permitted network. Enter the phone's current address in a browser, for example http://192.168.0.10. Use the management protocol and port configured on the device. If the web port is changed to 88, the corresponding example is http://192.168.0.10:88.

First-login security

The documented factory web credentials for this phone are username root and password root. A previously provisioned phone may use different credentials. Ask your administrator if the factory credentials do not work.

1. After signing in, open Password or the password-change control provided by your firmware.
2. Change the factory password to a unique password. If a guest or additional user account is enabled, change its default credentials or disable it when it is not required.
3. Use HTTPS for management where supported and restrict access to the trusted management network. Keep legacy HTTP, Telnet, FTP and TFTP services disabled when they are not required.
4. Sign out when configuration is complete. Store provisioning credentials, configuration backups and packet captures securely.

* HTTPS, password controls and service-disabling options depend on firmware. A different port number alone does not provide authentication or encryption. Do not expose the management interface directly to the public Internet.

PC port and management ports

The web PC Port page offers the supported bridge/router options. In bridge mode, the computer normally shares the phone's upstream network. Router mode requires a planned PC-side IP address, subnet and, where enabled, DHCP range. A SOCKS5 proxy is a separate network option; enter its server, port and credentials only if required by the administrator.

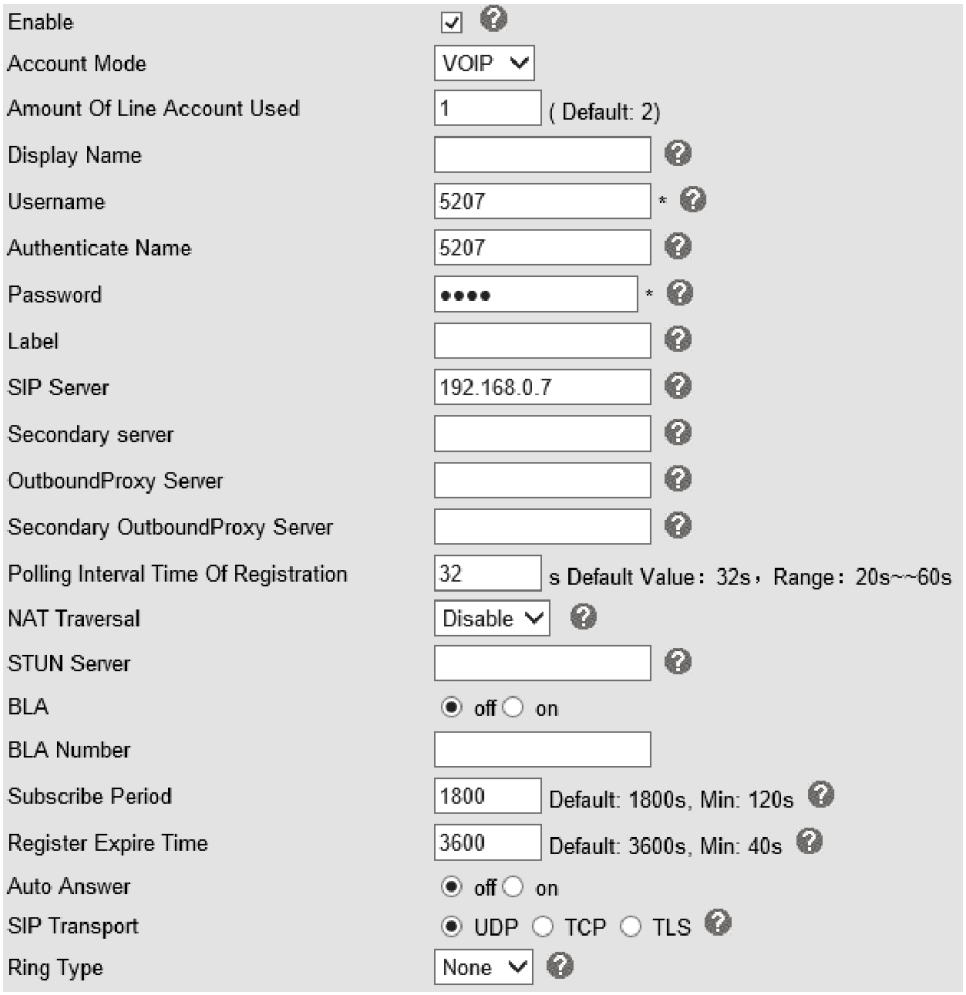
If Telnet is specifically required for maintenance, use the syntax expected by your client. For a command-line client that accepts separate host and port arguments, an example is telnet 192.168.0.200 2003. This differs from the colon notation used in a web URL. Disable legacy access after use.

5. Set up SIP service

On the phone, open System Settings > Advanced Settings > SIP > Account SIP, or use the web SIP Account page. Select the required account, enter the provider settings and enable it. Save and confirm registration before making a test call.

Setting	Description
Activate / Enable registration	Enable the selected SIP account after entering the required service settings.
Username / SIP user / Phone number	The registration name or telephone number assigned by the SIP service provider. Match the field names used by your firmware.
Authentication name and password	The credentials used to authenticate the SIP account. They may differ from the phone number.
Display name / Label	The name presented to the called party or shown locally for the account. Presentation also depends on the SIP service.
SIP server / Proxy address	The IP address or domain name of the provider's SIP server.
SIP server / Proxy port	The server port specified by the provider. Use the value required by the selected transport; do not assume every service uses the same port.
Outbound proxy / Backup proxy	Enter these only if supplied by the provider. A backup server is used when supported and when the primary service is unavailable.
Realm / Domain	Enter the provider's SIP domain or authentication realm when required.
Transport	Select UDP, TCP or TLS as supported by both phone and service. TLS also requires the appropriate certificate configuration.
Registration status	Save or apply the settings and check the registration status text. Use Registered or the account icon rather than relying on color alone.

* A SIP account is a registration identity; a call appearance or line key represents a call or an account shortcut. Their counts are not interchangeable. Use only the account and key slots available in your firmware.



SIP account configuration (cropped). Example values only; use the settings assigned to your device and service.*

Advanced SIP parameters*

Change advanced parameters only when directed by your SIP administrator. Unsupported options may be absent from the interface.

Setting	Description
Registration expiration / Retry interval	Controls registration renewal and retry timing. Use the interval required by the SIP server.
STUN / NAT traversal	Configure the STUN server, port and binding interval when the network design requires STUN. A successful STUN exchange does not replace correct firewall and SIP settings.
DNS SRV	Allows a provider domain to be resolved to SIP service records when supported.
Keep alive / rport	Helps maintain NAT mappings and response routing. Select the packet type and interval required by the deployment.
DTMF type	Choose in-band, RTP telephone-event (RFC 2833), SIP INFO or Auto as required by the service. Configure the telephone-event payload value to match the server.

Setting	Description
Codecs / Response single codec	Set the codec preference order. A call requires a codec supported at both ends. Single-codec response limits the codecs offered in a response when supported.
Anonymous calls / Anonymous-call rejection	Controls caller-identity presentation and rejection of calls without caller identity. Provider support is required.
Feature codes / Feature synchronization	Uses provider-defined dial codes or server synchronization for DND, forwarding and other functions. Enter only codes supplied by your administrator.
Voicemail subscription / MWI	Subscribe to message-waiting notifications and set the retrieval number. MWI reports waiting messages; it does not automatically forward every call to voicemail.
Conference type / Server conference number	Select local mixing or a provider-hosted conference. For a hosted conference, enter the room number supplied by the provider.
Transfer timeout	Sets how long a transfer operation may remain pending.
Session timer	Controls SIP session refreshes. Choose values compatible with the server; a failed refresh may end a call.
Strict proxy / Contact / User Agent	Advanced interoperability settings for routing, Contact headers and device identification. Retain defaults unless the service requires a change.
SIP protocol version / Caller ID header	Use the server-compatible SIP version and caller-identity header. Legacy protocol modes should be used only when required by the deployment.
PRACK / GRUU / SCA / BLA / BLF list	Enables supported SIP capabilities such as reliable provisional responses, globally routable identifiers, shared/bridged call appearance and busy-lamp lists. These require compatible server features.
SIP/RTP encryption and certificates	Use the security mode supported by the phone and the SIP service. Do not equate a proprietary or legacy encryption option with TLS or SRTP; verify the mode with the administrator.
Use VPN	Routes supported signaling or account traffic through the configured VPN when the deployment requires it.

6. Everyday calls

Make and answer calls

1. Lift the handset, activate the speakerphone, or select the headset if one is connected and supported.
2. Select an available account or line if needed, enter the number and press Send, Dial or the configured send key. You can also select a contact, history entry or configured speed-dial key.
3. For an incoming call, lift the handset or select the speakerphone, headset or Answer control. To decline the call, select Reject.
4. To finish, return the handset to its cradle or use End. In speakerphone or headset mode, use the corresponding control to end the active call.

Mute, hold and resume

Press Mute during a call to stop sending microphone audio. You can still hear the other party. Press Mute again to restore the microphone. Use Hold to pause a call and Resume or Unhold to return to it. If several calls are displayed, select the intended call before resuming or ending it.

* A held call remains active until it is resumed and ended, transferred or cleared by the service. Confirm that the call has ended before leaving the phone.

Call waiting and redial

When call waiting is enabled, another incoming call can be presented while you are talking. Answering it normally places the current call on hold. Select the intended call to switch between calls. Use Redial to call the last dialed number, or open History to choose an earlier number.

Do Not Disturb and forwarding

DND rejects or suppresses incoming calls according to the phone and provider configuration. Enable or disable it using the DND key or the Features menu. Forwarding sends incoming calls to another number. Select Always, Busy or No Answer, enter the destination, and save. For No Answer, also set the delay. Disable forwarding when it is no longer required.

* DND, forwarding, call waiting and voicemail may be controlled by the SIP server. A phone-level setting and an account-level setting can have different scope. Emergency dialing also depends on service availability and network power; configure emergency numbers with your administrator.

Transfer a call

1. For a blind transfer, select Transfer during the call, enter the destination and complete the transfer using the displayed Transfer/Send control or the configured transfer method. You do not speak to the recipient first.
2. For an attended transfer, select Transfer, dial the destination and wait for the recipient to answer. Speak to the recipient, then select Transfer again to connect the two other parties.
3. For a semi-attended transfer, complete the transfer while the destination is ringing, if supported. If the destination cannot be reached, cancel the attempt and resume the original call.

* Transfer behavior, on-hook transfer and send-key actions depend on the dial-plan settings and SIP service. Use the action displayed by your firmware; the server must support the required transfer method.

Conference calls

1. During an active call, select Conf, Conference or New Call. The existing call is placed on hold.
2. Dial the additional participant and wait for an answer.
3. Select Conf or the displayed Join/Conference control if the participants have not already been joined. Verify that the active participants appear in the conference.
4. Use Split to separate the calls where supported, or End to leave or end the conference according to the service configuration.

* Local conference size and whether ending the host call disconnects all participants depend on model, firmware and provider settings. A server-hosted conference may have different limits.

7. Contacts, messages and programmable keys

Phonebook and call history

Open Phonebook, Contacts or Directory to search local contacts. To add a contact, select Add, enter the name and telephone number, choose a group if required, and save. To edit or remove an entry, select it and use the displayed edit or delete control. Confirm the intended selection before deleting.

History contains incoming, outgoing and missed calls. Select an entry to dial it, view details, save it as a contact or add it to the block list. Delete All or Clear may remove an entire list. Export contacts before bulk deletion if they must be retained.

Remote directories and LDAP*

A remote XML phonebook retrieves a directory from an administrator-managed server. Enter the phonebook name and URL, together with authentication credentials when required. LDAP additionally requires a server address and port, search base, search filter or attribute mappings, and any authentication settings. Obtain these values from your directory administrator; do not copy addresses or credentials from an illustration.

Groups and blocked numbers

Groups organize contacts. Creating or deleting a group does not necessarily add or remove the contacts themselves. A block list rejects calls from selected numbers or prefixes. Add only the entries you intend to block, and remove them to accept calls again. Where outgoing-call barring or an allow list is offered, review the rule scope before enabling it.

Voicemail and messages

A message-waiting indicator shows that the service has reported voicemail. Press the message key or dial the configured retrieval number and follow the voice prompts. The retrieval number and password come from the service provider. Where text messaging is supported, use Messaging or SMS to read, compose, send and delete messages.

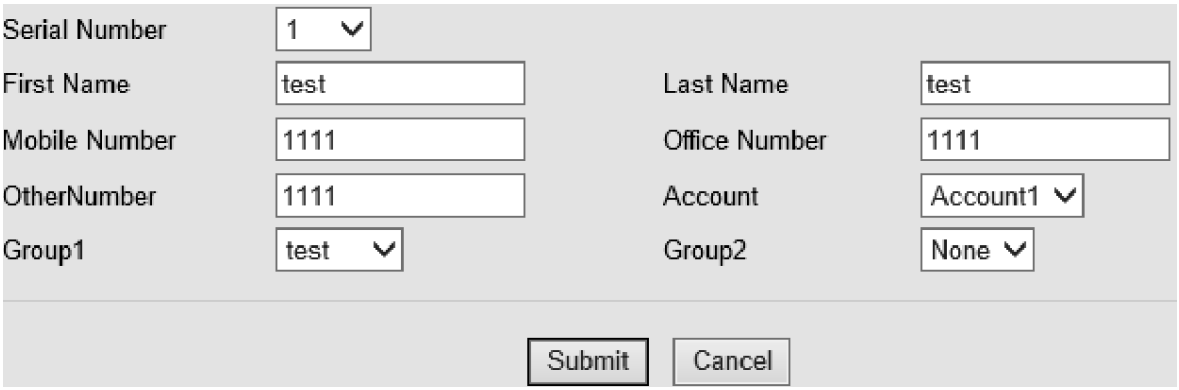
Programmable keys*

Open the phone's key settings or the web Function Key page. Select an available key, choose a type, enter the required value and label, then save. The number of physical keys and the number of assignable key slots or pages are different; configure only the slots available on the device.

Setting	Description
Line	Selects the assigned SIP account or opens its dialer.
Speed dial / Prefix	Dials a stored destination, or inserts a configured prefix before the number.
BLF	Monitors the status of a configured extension. Pickup or one-touch transfer also requires the server's pickup/transfer configuration.
Presence	Displays the availability reported by a compatible presence service. Avoid assigning BLF and Presence to the same monitored number unless the deployed firmware and server explicitly support it.
Intercom	Places an intercom call when permitted by the receiving phone and service.
MWI / Voicemail	Opens voicemail or dials the configured retrieval number.
DTMF	Sends a configured sequence of digits during a call.
Key event	Assigns an available action such as DND, Hold, Transfer, Redial, Headset, Pickup or Lock.
URL / Remote phonebook	Opens the configured URL or supported directory action.
Multicast	Sends an RTP paging stream to the configured multicast destination when supported.

* Directory formats, storage capacity, key types and subscription behavior vary with firmware. BLF, Presence, pickup and shared-line features require a compatible SIP service.

Contact editor example



Local contact editor (cropped). Example values only; use the settings assigned to your device and service.*

8. Phone preferences

Phone menu and web settings

Use System Settings > Phone Settings for the language, ring, volume and available date/time settings. Use the web interface for settings that are absent from the keypad menu, such as a custom ringtone upload, firmware management or advanced dial-plan options. Do not assume that all time settings are web-only: the described phone menu includes date/time controls.

Time, display and language

Select the appropriate time zone and daylight-saving rule before checking the displayed time. For automatic time, configure the primary and optional secondary SNTP server and the resynchronization interval, or use network-supplied settings where supported. For manual time, disable automatic synchronization as required, enter the date and time, and save. Choose the date format and 12- or 24-hour clock separately.

Use Display, Screen or Basic Settings to adjust the available backlight, contrast, wallpaper and idle-screen options. Choose the language from the list provided by the installed firmware. The languages offered by the LCD and web interface may differ. Reconnect or restart only if the interface requests it.

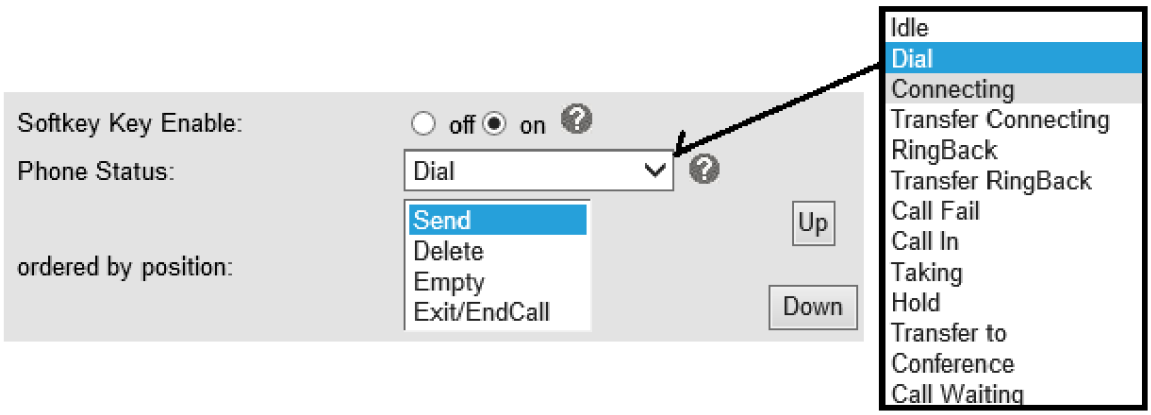
* Available language packs, screen controls, image formats and upload limits depend on firmware. Follow the requirements shown in the upload page. A filename extension must remain contiguous, for example TITLE_LOGO.bmp, without a space after the period.

Audio settings

Setting	Description
Codec preference	The first through sixth codec fields, where present, set the preferred order. Use only codecs supported by the SIP service; disable unused lower-priority entries if required.
Handset volume	Adjusts the audio heard through the handset.
Speakerphone volume	Adjusts the hands-free speaker level. Lower the volume if acoustic feedback or echo occurs.
Headset volume / Ring from headset	Adjusts headset audio and, where supported, selects the headset as the ringtone destination. This is distinct from the handset.
Ring volume / Ring type	Selects the ringtone and its level. Custom ringtone format and size limits are firmware-dependent.
Microphone gain / Offset	Adjusts the selected microphone path. Excessive gain can cause noise, distortion or echo.
DTMF payload type	The RTP telephone-event payload identifier must match the SIP service. This is an identifier, not an audio-frame duration or a timestamp setting.
G.729 / G.723.1 / G.722 parameters	Controls codec-specific payload length, bit rate or timestamp behavior. Use values compatible with the provider; retain defaults unless instructed otherwise.
VAD / Comfort noise	Voice activity detection can reduce transmission during silence; compatible comfort-noise handling is needed for natural call audio.
Jitter buffer	Balances tolerance of packet-delay variation against additional audio delay. Adaptive and fixed modes are offered by some firmware.
Echo cancellation / Side tone	Controls acoustic/line echo processing and local microphone feedback where supported.
On-hook / Hook-flash timing	Configures supported hook detection timing. Use the defaults or the values specified by the administrator.

Function keys and soft keys

Use the web Line Keys, Function Keys and Soft Key pages to assign the available controls. Soft-key functions can differ between idle, dialing and active-call screens. Select the context first, assign a supported action and save. A function cannot be invoked by a hardware key that the device does not have.



Context-dependent soft-key assignments (cropped). Example values only; use the settings assigned to your device and service.*

9. Dial plans and call-feature settings

Dial-plan rules change when and where a number is sent. Make changes with your SIP administrator and test both ordinary and required emergency destinations after applying a rule.

Setting	Description
Press # to send	Sends the entered number when # is pressed, where enabled. Consider services that need # as part of a dial string.
Fixed length / Dial length	Sends a number when the configured digit count is reached. Set the length for the actual numbering plan.
No-dial timeout / Send after seconds	Sends the number after the configured interval without additional digits.
Blind transfer / On-hook transfer	Changes which send or hang-up action completes a transfer. Enable only the method expected by the users and service.
Attended transfer on conference on-hook	May connect the remaining conference participants when the host hangs up. Confirm this behavior before relying on it.
E.164 / Dial prefix	Where supported, formats numbers or adds a configured outside-line prefix. Do not assume that the displayed number and transmitted number are identical.
Dial peer / Dial rule	Matches a full number or a prefix and can substitute, add or remove digits or select a destination. Check the entire resulting number before saving.

Dial-peer fields and examples

Setting	Description
Number / Match	The full number or prefix to match. Prefix matching may use a trailing T; wildcard syntax is specific to the displayed interface.
Destination / Port	The target IP address or domain and signaling port. Leave optional fields unset unless required by the deployment.
Alias: all	Replaces the matched number with the configured value. For example, a full match of 32 can map to 833333 without also matching 322.
Alias: add	Adds the configured prefix. For example, adding 007 to 9312 produces 0079312.
Alias: rep	Replaces the configured leading digits. For example, an administrator can map a short access prefix to an area code.
Alias: del / Delete length	Removes the specified number of leading digits. Deleting one leading digit from 98322 produces 8322; the condition concerns the start of the number.
Suffix	Adds digits after the number. For example, adding 0088 to 1383322 produces 13833220088.
Wildcards / Timer	Use the exact symbols documented by the local rule editor. A digit wildcard, a bracketed range, a variable-length match and a send timer may have different meanings in dial-peer and digit-map interfaces.

* Match syntax and precedence depend on the firmware page. Do not transfer a rule unchanged between different rule editors without checking it. Use a private, administrator-supplied address for direct-IP calling.

Additional call features*

Setting	Description
Hotline	Automatically calls a configured destination after going off-hook and any configured delay. Read the unit and allowed range shown by the firmware before setting the timeout.
Auto answer / Intercom	Answers eligible calls automatically. Configure the delay, allowed source/group and audio device carefully to prevent unintended audio access.
Auto redial / Call completion	Retries or monitors a busy destination according to supported server/phone behavior. Configure retry count and interval where offered.
Pickup / Call park	Uses provider-defined codes or keys to answer or retrieve a call from another extension. Obtain the required codes from the administrator.
Password dial / Hide DTMF	Masks displayed digits according to prefix and length settings. Display masking does not encrypt signaling or media.
Pre-dial	Allows number entry before opening the audio channel. Send the number using the configured send action.
Action URL / Active URI	Reports phone events to an approved server or accepts supported control actions. Restrict control requests to trusted source addresses.
Push XML	Allows approved server content to be displayed or processed by the phone. Configure only trusted servers.
Keypad lock / Emergency number	Restricts local controls. Configure and test required emergency dialing with the service provider; a lock exception does not guarantee service availability.

10. Advanced networking and security

VPN and VLAN are separate settings

The VPN page sets the supported tunnel type, server address, username and password. The VLAN page sets the network segmentation and priority options. Configure these separately using values provided by the network administrator. A VPN option alone does not guarantee encrypted traffic; confirm the actual tunnel mode and security properties before use.

QoS and paging

The QoS page sets the supported signaling and voice priority values. The paging configuration defines multicast groups and destinations. Paging entries are ordered by the interface's priority rules; configure only destinations approved for the local network and test how pages interact with active calls before deployment.

Encryption and management

The legacy SIP/RTP Encryption page offers an RC4-based mode. Treat it as a legacy interoperability option and do not describe it as a substitute for modern transport/media security. Use a supported secure configuration agreed with the SIP administrator, or keep the phone within the approved protected network. The Password page manages the administrator and user web passwords.

11. Maintenance, recovery and support

Back up the configuration

Open Phone Maintenance > Configurations, Backup or the corresponding maintenance page. Export the configuration and, where available, the phonebook to a secure location before upgrading, resetting or making major changes. Backups may contain credentials or personal information. Import only a compatible backup for the correct model and firmware.

Upgrade firmware

1. Identify the model, hardware revision and current firmware in System Information and on the device label. Obtain a matching firmware package from D-Link support or your authorized administrator.
2. Read the package release notes, including any required upgrade sequence and whether settings or user data will be erased. Make a backup first.
3. Use the web Upgrade/Update page or the approved provisioning method. Select the exact file and start the update.
4. Keep power and the network connection stable throughout the update. Do not disconnect cables, switch off the phone or start a second update.
5. Wait until the phone has completed its restart. Reconnect, confirm the firmware version and verify network registration, calling and any required custom settings.

* Recovery key combinations, USB/SD package layout and wipe requirements are revision-specific. Do not use an unverified key combination or erase user data as a routine upgrade step. If the device will not boot, obtain the recovery procedure for the exact revision from D-Link support.

Automatic provisioning*

Provisioning allows an administrator to supply configuration or firmware from an approved server. The interface may provide DHCP options, SIP PnP, a static server URL, scheduled checks or TR-069/CWMP. Enter the server address, transfer method, authentication credentials and filenames supplied for your deployment. Configure the interval or boot-time check only as needed.

Setting	Description
Server / Protocol / Credentials	Specifies where the phone retrieves configuration or firmware. Use an encrypted supported method when available and protect credentials.
Configuration version / Digest	Controls whether a downloaded configuration is applied. A matching version or digest may suppress an unnecessary update.

Setting	Description
DHCP option / SIP PnP	Allows the network to advertise a provisioning server. Enable only on a trusted, controlled network.
Boot check / Scheduled check	Controls when the phone looks for updates. A separate option may temporarily disable normal use during a boot-time check; read the exact label before enabling it.
Download software / Configuration / Phonebook	Selects which supported items are retrieved. Expansion-module downloads apply only to compatible connected hardware.
Encryption key	Protects a supported configuration file. Obtain the key securely from the administrator; do not publish it.
CWMP / ACS	Sets the approved ACS URL, authentication and reporting interval where supported. Choose the server type required by the deployment; ordinary installations should not copy operator-specific example settings.

Restart or factory reset

Restart reboots the phone and is different from factory reset. Save configuration changes first, then use Reboot/Restart and wait for the phone to return to service. A factory reset removes user configuration and may erase contacts, preferences and other stored data. Export required data first. Use Factory Reset only when the administrator has the information needed to provision the phone again. Read and confirm the on-screen warning before proceeding.

Logs and packet captures

For troubleshooting, use the supported System/Tools or maintenance page to collect a log or packet capture. Start capture, reproduce the problem, then stop and save the file. Captures and logs can contain telephone numbers, account data, media or authentication material. Share them only with authorized support personnel through an approved channel. Disable packet capture, mirroring and verbose logging after the investigation.

Troubleshooting

Setting	Description
Phone does not start	Check the power adapter rating and connections, or the PoE switch and cable where PoE is supported. Try an approved power source. Do not attempt repeated firmware recovery without the correct procedure.
Cannot obtain an IP address	Check that the uplink is in the network-facing port. Verify the switch link, DHCP service and VLAN. For static settings, confirm the address, subnet, gateway and DNS values.
SIP account does not register	Check network access, time, server address, account credentials and required transport/certificates. Review the registration-status text and contact the SIP provider if needed.
No audio or one-way audio	Check the handset/headset connection, mute state, selected audio device, codecs, RTP path, firewall/NAT rules and available bandwidth.
Echo or interrupted speakerphone audio	Reduce speaker volume and check microphone placement and network quality. Ask the administrator to review audio settings if the problem persists.
Web interface unavailable	Use the current IP address and configured management protocol/port. Check whether HTTP was disabled or the phone moved to a different VLAN. Do not bypass certificate warnings without administrator guidance.
Settings differ from this guide	Check the hardware revision and firmware. Menu names, available parameters and defaults can vary. Follow the approved documentation for the exact device.

D-Link support

For documentation, firmware and assistance, visit dlink-mea.com. Identify the model and hardware revision from the device label before downloading files. Use only a firmware package approved for that exact model and revision.

When requesting help, provide the model and hardware revision, firmware version, a description of the problem, the relevant network/service configuration and the steps that reproduce it. Do not include passwords, private keys or unredacted contact information in a public support request.