CHAPTER 21
VoIP

21.1 Overview

You can make calls over the Internet using VoIP technology. For this, you first need to set up a SIP account with a SIP service provider.

Use this chapter to:

- Connect an analog phone to the Zyxel Device.
- Configure settings such as speed dial.
- Configure network settings to optimize the voice quality of your phone calls.

21.1.1 What You Can Do in this Chapter

These screens allow you to configure your Zyxel Device to make phone calls over the Internet and your regular phone line, and to set up the phones you connect to the Zyxel Device.

- Use the **SIP Account** screen (Section 21.3 on page 229) to set up information about your SIP account, control which SIP accounts the phones connected to the Zyxel Device use and configure audio settings such as volume levels for the phones connected to the Zyxel Device.
- Use the **SIP Service Provider** screen (Section 21.4 on page 233) to configure the SIP server information, QoS for VoIP calls, the numbers for certain phone functions, and dialing plan.
- Use the **Phone Device** screen (Section 21.5 on page 238) to control which SIP account(s) each phone uses to handle outgoing and incoming calls.
- Use the **Region** screen (Section 21.6 on page 241) to change settings that depend on the country you are in.
- Use the **Call Rule** screen (Section 21.7 on page 241) to set up shortcuts for dialing frequently-used (VoIP) phone numbers.
- Use the **Call History** screen (Section 21.8 on page 242) to view detailed information for each outgoing call you made or each incoming call from someone calling you. You can also view the summary list of received, dialed and missed calls.

You don't necessarily need to use all these screens to set up your account. In fact, if your service provider did not supply information on a particular field in a screen, it is usually best to leave it at its default setting.

21.1.2 What You Need to Know About VoIP

VoIP

VoIP stands for Voice over IP. IP is the Internet Protocol, which is the message-carrying standard the Internet runs on. So, Voice over IP is the sending of voice signals (speech) over the Internet (or another network that uses the Internet Protocol).

SIP

SIP stands for Session Initiation Protocol. SIP is a signaling standard that lets one network device (like a computer or the Zyxel Device) send messages to another. In VoIP, these messages are about phone calls over the network. For example, when you dial a number on your Zyxel Device, it sends a SIP message over the network asking the other device (the number you dialed) to take part in the call.

SIP Accounts

A SIP account is a type of VoIP account. It is an arrangement with a service provider that lets you make phone calls over the Internet. When you set the Zyxel Device to use your SIP account to make calls, the Zyxel Device is able to send all the information about the phone call to your service provider on the Internet.

Strictly speaking, you don't need a SIP account. It is possible for one SIP device (like the Zyxel Device) to call another without involving a SIP service provider. However, the networking difficulties involved in doing this make it tremendously impractical under normal circumstances. Your SIP account provider removes these difficulties by taking care of the call routing and setup - figuring out how to get your call to the right place in a way that you and the other person can talk to one another.

SIP Address

A SIP address is a URI (Uniform Resource Identifier) that resembles an email address, using the format: user@domain. It uniquely identifies a telephone extension over a VoIP system. A SIP address of 123-45-67@voip-provider.net tells a client to connect to voip-provider.net and request a connection to 123-45-67. While VoIP can only send voice messages over the Internet, SIP (though strictly speaking is a type of VoIP) can send voice, data, video, and other media. VoIP phones also need to be connected to a computer to function, whereas SIP phones only need to be connected to a modem.

How to Find Out More

~~See Chapter 4 on page 37 for a tutorial showing how to set up these screens in an example scenario.~~

See Section 21.10 on page 244 for advanced technical information on SIP.

21.2 Before You Begin

- Before you can use these screens, you need to have a VoIP account already set up. If you don't have one yet, you can sign up with a VoIP service provider over the Internet.
- You should have the information your VoIP service provider gave you ready, before you start to configure the Zyxel Device.

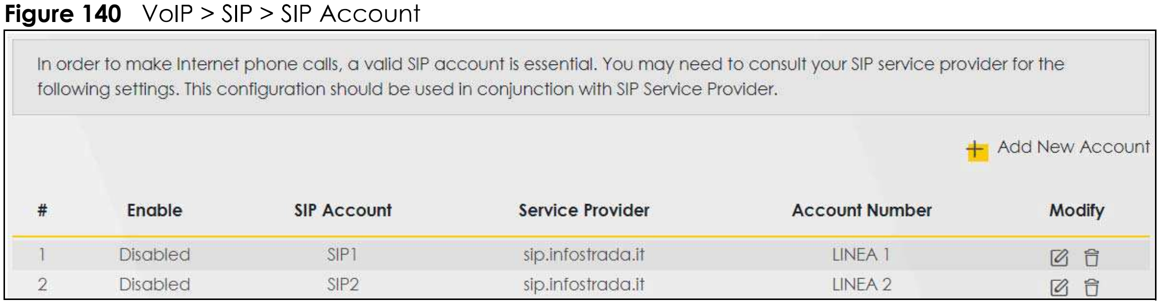
21.3 The SIP Account Screen

The Zyxel Device uses a SIP account to make outgoing VoIP calls and check if an incoming call's destination number matches your SIP account's VoIP number. In order to make or receive a VoIP call, you need to enable and configure a SIP account and map it to a phone port. The SIP account contains information that allows your Zyxel Device to connect to your VoIP service provider.

See Section 21.3.1 on page 229 for how to map a SIP account to a phone port.

Use this screen to view SIP account information. You can also enable and disable each SIP account. To access this screen, click **VoIP > SIP > SIP Account**.

Figure 140 VoIP > SIP > SIP Account



Each field is described in the following table.

Table 96 VoIP > SIP > SIP Account

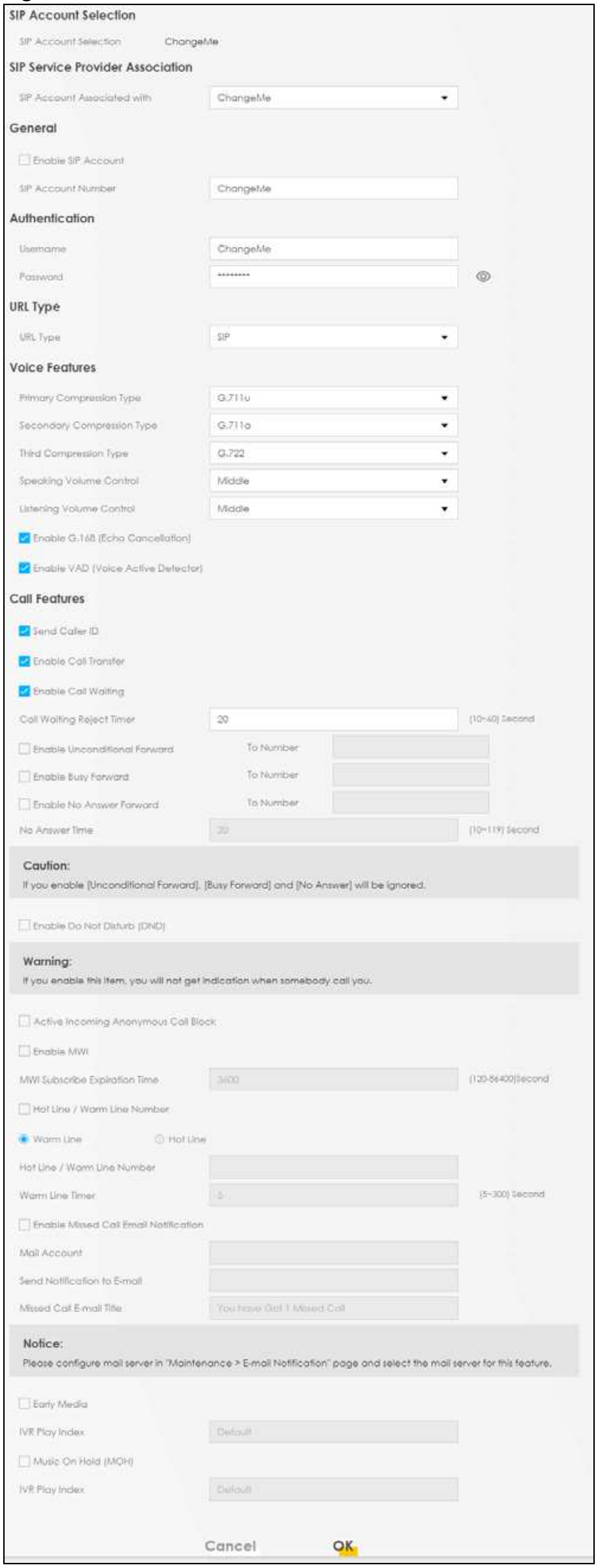
LABEL	DESCRIPTION
Add new account	Click this to configure a SIP account.
#	This is the index number of the entry.
Enable	This shows whether the SIP account is activated or not. ~~A yellow bulb signifies that this SIP account is activated. A gray bulb signifies that this SIP account is not activated.~~
SIP Account	This shows the name of the SIP account.
Service Provider	This shows the name of the SIP service provider.
Account Number.	This shows the SIP address.
Modify	Click the **Edit** icon to configure the SIP account. Click the **Delete** icon to delete this SIP account from the Zyxel Device.

21.3.1 The SIP Account Add/Edit Screen

Use this screen to configure a SIP account and map it to a phone port in the **Phone Device** screen. To access this screen, click the **Add New Account** button or click the **Edit** icon of an entry in the **VoIP > SIP > SIP Account** screen.

Note: ~~Click **more** to see all the fields in the screen.~~ You don't necessarily need to use all these fields to set up your account. ~~Click **less** to see and configure only the fields needed for this feature.~~

Figure 141 VoIP > SIP > SIP Account > Add new account/Edit



Each field is described in the following table.

Table 97 VoIP > SIP > SIP Account > Add new account/Edit

LABEL	DESCRIPTION
SIP Account Selection	
SIP Account Selection	This field displays ~~ADD_NEW~~**ChangeMe** if you are creating a new SIP account or the SIP account you are modifying.
SIP Service Provider Association	
SIP Account Associated with	Select the SIP service provider profile to use for the SIP account you are configuring in this screen. You should already have configured a SIP service provider profile in the **SIP Service Provider** screen. This field is read-only when you are modifying an existing SIP account.
General	
Enable SIP Account	Select this if you want the Zyxel Device to use this account. Clear it if you do not want the Zyxel Device to use this account.
SIP Account Number	Enter your SIP address. In the full SIP URI, this is the part before the @ symbol. You can use up to 127 printable ASCII characters.
Authentication	
Username	Enter the user name for registering this SIP account, exactly as it was given to you. You can use up to 95 printable ASCII characters.
Password	Enter the user name for registering this SIP account, exactly as it was given to you. You can use up to 95 printable ASCII Extended set characters.
URL Type	
URL Type	Select whether or not to include the SIP service domain name when the Zyxel Device sends the SIP address. **SIP** - include the SIP service domain name. **TEL** - do not include the SIP service domain name.
Voice Features	
Primary Compression Type Secondary Compression Type Third Compression Type	Select the type of voice coder/decoder (codec) that you want the Zyxel Device to use. G.711 provides high voice quality but requires more bandwidth (64 kbps). G.711 is the default codec used by phone companies and digital handsets. • **G.711a** is typically used in Europe. • **G.711u** is typically used in North America and Japan. **G.726-24** operates at **24** kbps. **G.726-32** operates at **32** kbps. By contrast, **G.729** only requires 8 kbps. **G.722** is a 7 KHz wideband voice codec that operates at 48, 56 and 64 kbps. By using a sample rate of 16 kHz, G.722 can provide higher fidelity and better audio quality than narrowband codecs like G.711, in which the voice signal is sampled at 8 KHz. The Zyxel Device must use the same codec as the peer. When two SIP devices start a SIP session, they must agree on a codec. Select the Zyxel Device's first choice for voice coder/decoder. Select the Zyxel Device's second choice for voice coder/decoder. Select **None** if you only want the Zyxel Device to accept the first choice. Select the Zyxel Device's third choice for voice coder/decoder. Select **None** if you only want the Zyxel Device to accept the first or second choice.

Table 97 VoIP > SIP > SIP Account > Add new account/Edit (continued)

LABEL	DESCRIPTION
Speaking Volume Control	Select the loudness that the Zyxel Device uses for speech that it sends to the peer device.
Listening Volume Control	Select the loudness that the Zyxel Device uses for speech that it receives from the peer device.
Enable G.168 (Echo Cancellation)	Select this if you want to eliminate the echo caused by the sound of your voice reverberating in the telephone receiver while you talk.
Enable VAD (Voice Active Detector)	Select this if the Zyxel Device should stop transmitting when you are not speaking. This reduces the bandwidth the Zyxel Device uses.
Call Features	
Send Caller ID	Select this if you want to send identification when you make VoIP phone calls. Clear this if you do not want to send identification.
Enable Call Transfer	Select this to enable call transfer on the Zyxel Device. This allows you to transfer an incoming call (that you have answered) to another phone.
Enable Call Waiting	Select this to enable call waiting on the Zyxel Device. This allows you to place a call on hold while you answer another incoming call on the same telephone number.
Call Waiting Reject Timer	Specify the time in seconds that the Zyxel Device waits before rejecting the second call if you do not answer it.
Enable Unconditional Forward	Select this if you want the Zyxel Device to forward all incoming calls to the specified phone number. Specify the phone number in the **To Number** field on the right.
Enable Busy Forward	Select this if you want the Zyxel Device to forward incoming calls to the specified phone number if the phone port is busy. Specify the phone number in the **To Number** field on the right. If you have call waiting, the incoming call is forwarded to the specified phone number if you reject or ignore the second incoming call.
Enable No Answer Forward	Select this if you want the Zyxel Device to forward incoming calls to the specified phone number if the call is unanswered. (See **No Answer Time**.) Specify the phone number in the **To Number** field on the right.
No Answer Time	This field is used by the **Active No Answer Forward** feature. Enter the number of seconds the Zyxel Device should wait for you to answer an incoming call before it considers the call unanswered.
Enable Do Not Disturb	Select this to set your phone to not ring when someone calls you.
Active Incoming Anonymous Call Block	Select this if you do not want the phone to ring when someone tries to call you with caller ID deactivated.
Enable MWI	Select this if you want to hear a waiting (beeping) dial tone on your phone when you have at least one voice message. Your VoIP service provider must support this feature.
MWI Subscribe Expiration Time	Keep the default value of this field unless your VoIP service provider tells you to change it. Enter the number of seconds the SIP server should provide the message waiting service each time the Zyxel Device subscribes to the service. Before this time passes, the Zyxel Device automatically subscribes again.
Hot Line/ Warm Line Number	Select this to enable the hot line or warm line feature on the Zyxel Device.
Hot Line	Select this to have the Zyxel Device dial the specified hot line number immediately when you pick up the telephone.

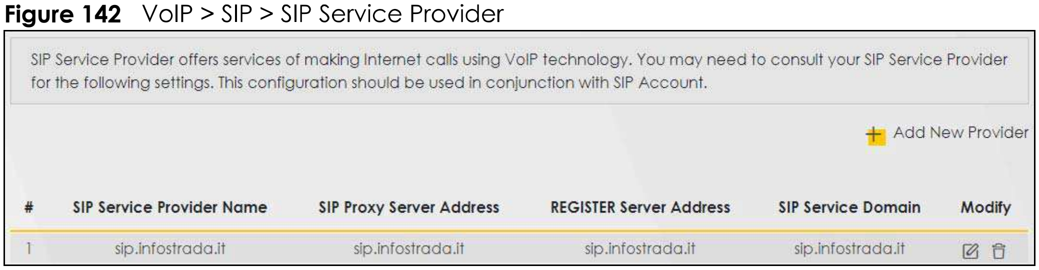
Table 97 VoIP > SIP > SIP Account > Add new account/Edit (continued)

LABEL	DESCRIPTION
Warm Line	Select this to have the Zyxel Device dial the specified warm line number after you pick up the telephone and do not press any keys on the keypad for a period of time.
Hot Line / Warm Line Number	Enter the number of the hot line or warm line that you want the Zyxel Device to dial.
Warm Line Timer	Enter a number of seconds that the Zyxel Device waits before dialing the warm line number if you pick up the telephone and do not press any keys on the keypad.
Enable Missed Call Email Notification	Select this option to have the Zyxel Device email you a notification when there is a missed call.
Mail Account	Select a mail account for the email address specified below. If you select **None** here, email notifications will not be sent via email. You must have configured a mail account already in the **Email Notification** screen.
Send Notification to Email	Notifications are sent to the email address specified in this field. If this field is left blank, notifications will not be sent via email.
Missed Call Email Title	Type a title that you want to be in the subject line of the email notifications that the Zyxel Device sends.
Early Media	Select this if you want people to hear a customized recording when they call you.
IVR Play Index	Select the tone you want people to hear when they call you. This field is configurable only when you select **Early Media**. See Section 21.10 on page 244 for information on how to record these tones.
Music On Hold (MOH)	Select this to play a customized recording when you put people on hold.
IVR Play Index	Select the tone to play when you put someone on hold. This field is configurable only when you select **Music on Hold**, See Section 21.10 on page 244 for information on how to record these tones.
Cancel	Click **Cancel** to exit this screen without saving.
OK	Click **OK** to save your changes.

21.4 The SIP Service Provider Screen

Use this screen to view the SIP service provider information on the Zyxel Device. A SIP provider offers Internet call services using VoIP technology. You may need to consult your SIP service provider for the following settings. Click **VoIP > SIP > SIP Service Provider** to open the following screen.

Figure 142 VoIP > SIP > SIP Service Provider



Each field is described in the following table.

Table 98 VoIP > SIP > SIP Service Provider

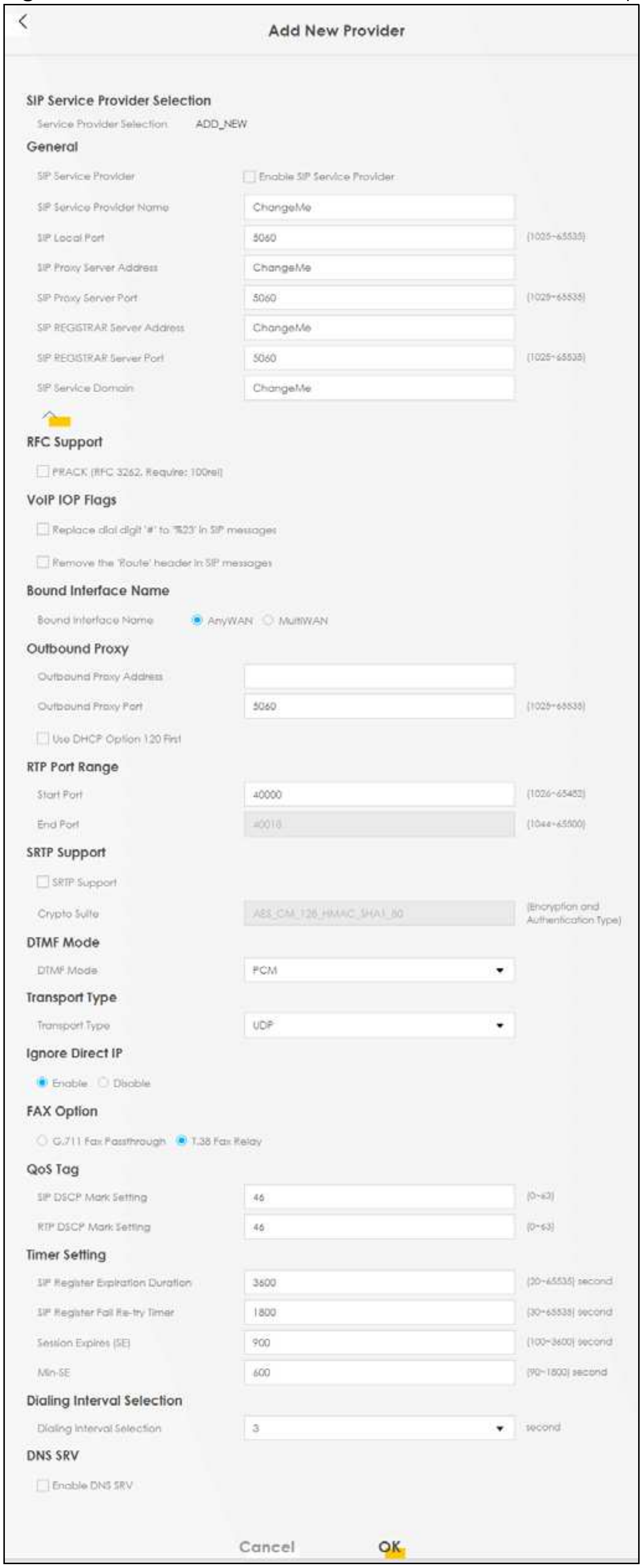
LABEL	DESCRIPTION
Add New Provider	Click this button to add a new SIP service provider.
#	This is the index number of the entry.
SIP Service Provider Name	This shows the name of the SIP service provider.
SIP Proxy Server Address	This shows the IP address or domain name of the SIP server.
REGISTER Server Address	This shows the IP address or domain name of the SIP register server.
SIP Service Domain	This shows the SIP service domain name.
Modify	Click the **Edit** icon to configure the SIP service provider. Click the **Delete** icon to delete this SIP service provider from the Zyxel Device.

21.4.1 The SIP Service Provider Add/Edit Screen

Use this screen to configure a SIP service provider on the Zyxel Device. Click the **Add New Provider** button or an **Edit** icon in the **VoIP > SIP > SIP Service Provider** to open the following screen.

Note: Click this ( ) to see all the fields in the screen. You don't necessarily need to use all these fields to set up your account. Click again to see and configure only the fields needed for this feature.

Figure 143 VoIP > SIP > SIP Service Provider > Add new provider/Edit



Each field is described in the following table.

Table 99 VoIP > SIP > SIP Service Provider > Add new provider/Edit

LABEL	DESCRIPTION
SIP Service Provider Selection	
Service Provider Selection	This field displays **ADD_NEW** if you are creating a new SIP service provider profile or the SIP service provider name you are modifying.
General	
SIP Service Provider	Select **Enable SIP Service Provider** to enable the SIP service provider.
SIP Service Provider Name	Enter the name of your SIP service provider.
SIP Local Port	Enter the Zyxel Device's listening port number, if your VoIP service provider gave you one. Otherwise, keep the default value.
SIP Proxy Server Address	Enter the IP address or domain name of the SIP server provided by your VoIP service provider. You can use up to 95 printable ASCII characters. It does not matter whether the SIP server is a proxy, redirect or register server.
SIP Proxy Server Port	Enter the SIP server's listening port number, if your VoIP service provider gave you one. Otherwise, keep the default value.
SIP REGISTRAR Server Address	Enter the IP address or domain name of the SIP register server, if your VoIP service provider gave you one. Otherwise, enter the same address you entered in the **SIP Server Address** field. You can use up to 95 printable ASCII characters.
SIP REGISTRAR Server Port	Enter the SIP register server's listening port number, if your VoIP service provider gave you one. Otherwise, enter the same port number you entered in the **SIP Server Port** field.
SIP Service Domain	Enter the SIP service domain name. In the full SIP URI, this is the part after the @ symbol. You can use up to 127 printable ASCII Extended set characters.
RFC Support	
PRACK (RFC 3262, Require: 100rel)	PRACK (RFC 3262) defines a mechanism to provide reliable transmission of SIP provisional response messages, which convey information on the processing progress of the request. This uses the option tag 100rel and the Provisional Response ACKnowledgement (PRACK) method. Select this to have the peer device require the option tag 100rel to send provisional responses reliably.
VoIP IOP Flags	Select the VoIP inter-operability settings you want to activate.
Replace dial digit '#' to '%23' in SIP messages	Replace a dial digit "#" with "%23" in the INVITE messages.
Remove the 'Route' header in SIP messages	Remove the 'Route' header in SIP packets.
Bound Interface Name	
Bound Interface Name	If you select **Any_WAN**, the Zyxel Device automatically activates the VoIP service when any LAN or WAN connection is up. If you select **Multi_WAN**, you also need to select two or more pre-configured WAN interfaces. The VoIP service is activated only when one of the selected WAN connections is up.
Outbound Proxy	
Outbound Proxy Address	Enter the IP address or domain name of the SIP outbound proxy server if your VoIP service provider has a SIP outbound server to handle voice calls. This allows the Zyxel Device to work with any type of NAT router and eliminates the need for STUN or a SIP ALG. Turn off any SIP ALG on a NAT router in front of the Zyxel Device to keep it from re-translating the IP address (since this is already handled by the outbound proxy server).
Outbound Proxy Port	Enter the SIP outbound proxy server's listening port, if your VoIP service provider gave you one. Otherwise, keep the default value.

Table 99 VoIP > SIP > SIP Service Provider > Add new provider/Edit (continued)

LABEL	DESCRIPTION
Use DHCP Option 120 First	Select this to enable the SIP server via DHCP option 120.
RTP Port Range	
Start Port End Port	Enter the listening port number(s) for RTP traffic, if your VoIP service provider gave you this information. Otherwise, keep the default values. To enter one port number, enter the port number in the **Start Port** and **End Port** fields. To enter a range of ports, • enter the port number at the beginning of the range in the **Start Port** field. • enter the port number at the end of the range in the **End Port** field.
SRTP Support	
SRTP Support	When you make a VoIP call using SIP, the Real-time Transport Protocol (RTP) is used to handle voice data transfer. The Secure Real-time Transport Protocol (SRTP) is a security profile of RTP. It is designed to provide encryption and authentication for the RTP data in both unicast and multicast applications. The Zyxel Device supports encryption using AES with a 128-bit key. To protect data integrity, SRTP uses a Hash-based Message Authentication Code (HMAC) calculation with Secure Hash Algorithm (SHA)-1 to authenticate data. HMAC SHA-1 produces a 80 or 32-bit authentication tag that is appended to the packet. Both the caller and callee should use the same algorithms to establish an SRTP session.
Crypto Suite	Select the encryption and authentication algorithm set used by the Zyxel Device to set up an SRTP media session with the peer device. Select **AES_CM_128_HMAC_SHA1_80** or **AES_CM_128_HMAC_SHA1_32** to enable both data encryption and authentication for voice data. Select **AES_CM_128_NULL** to use 128-bit data encryption but disable data authentication. Select **NULL_CIPHER_HMAC_SHA1_80** to disable encryption but require authentication using the default 80-bit tag.
DTMF Mode	
DTMF Mode	Control how the Zyxel Device handles the tones that your telephone makes when you push its buttons. You should use the same mode your VoIP service provider uses. **RFC2833** - send the DTMF tones in RTP packets. **PCM** - send the DTMF tones in the voice data stream. This method works best when you are using a codec that does not use compression (like G.711). Codecs that use compression (like G.729 and G.726) can distort the tones. **SIP INFO** - send the DTMF tones in SIP messages.
Transport Type	
Transport Type	Select the transport layer protocol **UDP** or **TCP** (usually UDP) used for SIP.
Ignore Direct IP	Select **Enable** to have the connected CPE devices accept SIP requests only from the SIP proxy/ register server specified above. SIP requests sent from other IP addresses will be ignored.
FAX Option	This field controls how the Zyxel Device handles fax messages.
G711 Fax Passthrough	Select this if the Zyxel Device should use G.711 to send fax messages. ~~You have to also select which operating codec (**G.711Mulaw** or **G.711Alaw**) to use for encoding/decoding FAX data.~~ The peer devices must use the same settings.
T38 Fax Relay	Select this if the Zyxel Device should send fax messages as UDP or TCP/IP packets through IP networks. This provides better quality, but it may have inter-operability problems. The peer devices must also use T.38.
QoS Tag	

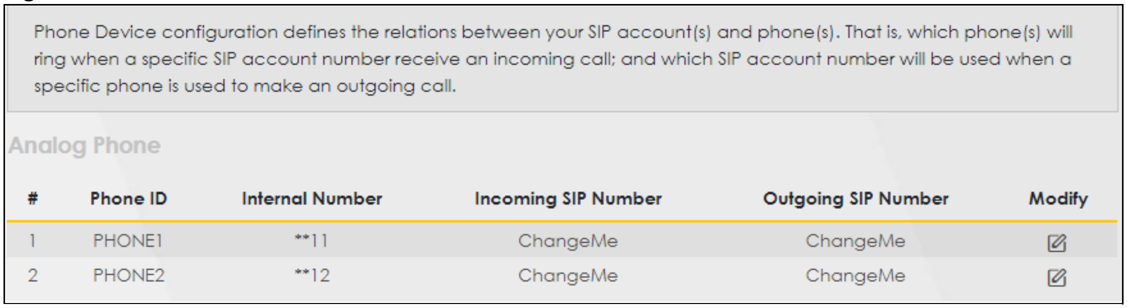
Table 99 VoIP > SIP > SIP Service Provider > Add new provider/Edit (continued)

LABEL	DESCRIPTION
SIP DSCP Mark Setting	Enter the DSCP (DiffServ Code Point) number for SIP message transmissions. The Zyxel Device creates Class of Service (CoS) priority tags with this number to SIP traffic that it transmits.
RTP DSCP Mark Setting	Enter the DSCP (DiffServ Code Point) number for RTP voice transmissions. The Zyxel Device creates Class of Service (CoS) priority tags with this number to RTP traffic that it transmits.
Timer Setting	
SIP Register Expiration Duration	Enter the number of seconds your SIP account is registered with the SIP register server before it is deleted. The Zyxel Device automatically tries to re-register your SIP account when one-half of this time has passed. (The SIP register server might have a different expiration.)
SIP Register Fail Re-try timer	Enter the number of seconds the Zyxel Device waits before it tries again to register the SIP account, if the first try failed or if there is no response.
Session Expires (SE)	Enter the number of seconds the Zyxel Device lets a SIP session remain idle (without traffic) before it automatically disconnects the session.
Min-SE	Enter the minimum number of seconds the Zyxel Device lets a SIP session remain idle (without traffic) before it automatically disconnects the session. When two SIP devices start a SIP session, they must agree on an expiration time for idle sessions. This field is the shortest expiration time that the Zyxel Device accepts.
Dialing Interval Selection	
Dialing Interval Selection	Enter the number of seconds the Zyxel Device should wait after you stop dialing numbers before it makes the phone call. The value depends on how quickly you dial phone numbers.
DNS SRV	
Enable DNS SRV	Select this to have the Zyxel Device use DNS procedures to resolve the SIP domain and find the SIP server's IP address, port number and supported transport protocol(s). The Zyxel Device first uses DNS Name Authority Pointer (NAPTR) records to determine the transport protocols supported by the SIP server. It then performs DNS Service (SRV) query to determine the port number for the protocol. The Zyxel Device resolves the SIP server's IP address by a standard DNS address record lookup. The **SIP Server Port** and **REGISTER Server Port** fields in the **General** section above are grayed out and not applicable and the **Transport Type** can also be set to **AUTO** if you enable this option.
Cancel	Click **Cancel** to exit this screen without saving.
OK	Click **OK** to save your changes.

21.5 The Phone Device Screen

Use this screen to view detailed information on phones used for Internet phone calls (SIP). You can define which phone(s) will ring when a specific SIP address receives an incoming call, and which SIP address will be used when an outgoing call is made with a specific phone. To access this screen, click **VoIP > Phone > Phone Device**.

Figure 144 VoIP > Phone > Phone Device



Each field is described in the following table.

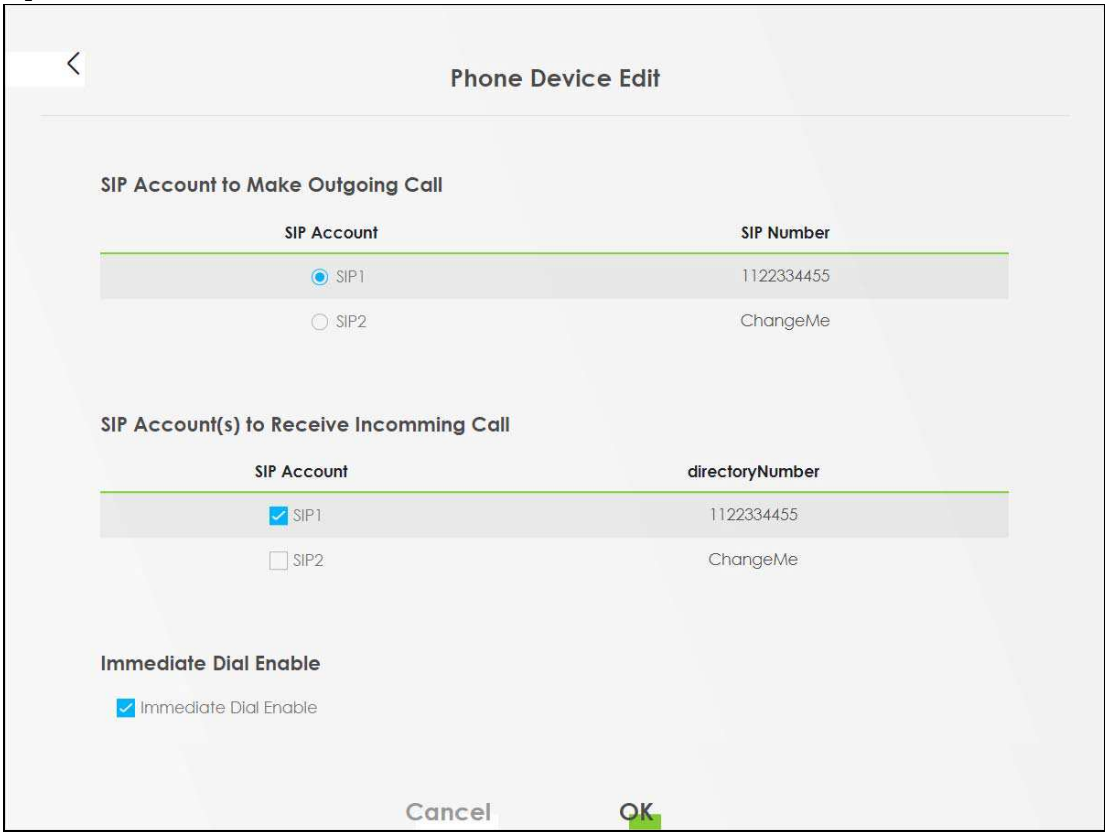
Table 100 VoIP > Phone > Phone Device

LABEL	DESCRIPTION
#	This displays the index number of the phone device.
Phone ID	This field displays the name of a phone port on the Zyxel Device.
Internal Number	This field displays the internal call prefix of a phone port on the Zyxel Device.
Incoming SIP Number	This field displays the SIP address that you use to receive calls on this phone port.
Outgoing SIP Number	This field displays the SIP address that you use to make calls on this phone port.
Modify	Click the **Edit** icon to configure the SIP account.

21.5.1 The Phone Device Edit Screen

Use this screen to control which SIP account(s) ~~and PSTN line~~ each phone uses. Click an **Edit** icon in **VoIP > Phone > Phone Device** to open the following screen.

Figure 145 VoIP > Phone > Phone Device > Edit



Each field is described in the following table.

Table 101 VoIP > Phone > Phone Device > Edit

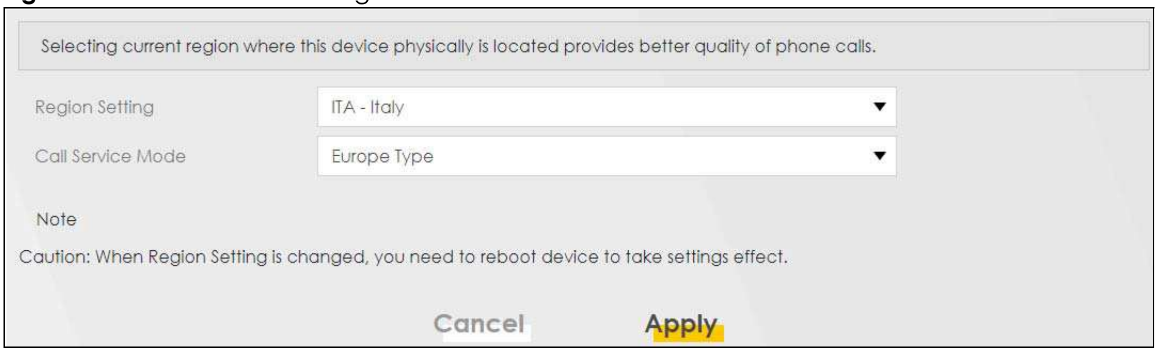
LABEL	DESCRIPTION
SIP Account to Make Outgoing Call	Select the SIP account you want to use when making outgoing calls with the analog phone connected to this phone port.
SIP Account(s) to Receive Incoming Call	Select a SIP account if you want to receive phone calls for the selected SIP account on this phone port. If you select more than one SIP account for incoming calls, there is no way to distinguish between them when you receive phone calls. If you do not select a source for incoming calls, you cannot receive any calls on this phone port.
Immediate Dial Enable	Select this if you want to use the pound key (#) to tell the Zyxel Device to make the phone call immediately, instead of waiting for the number of second you selected in the **Dialing Interval Selection** field of the **VoIP > SIP > SIP Service Provider > Add New Provider/Edit** screen. If you select this, dial the phone number, and then press the pound key. The Zyxel Device makes the call immediately instead of waiting. You can still wait, if you want.
Cancel	Click **Cancel** to exit this screen without saving
OK	Click **OK** to save your changes.

21.6 The Phone Region Screen

Use this screen to maintain settings that depend on which region of the world the Zyxel Device is in. Selecting the region where the device is physically located improves the quality of phone calls. To access this screen, click **VoIP > Phone > Region**.

Note: You need to reboot the device after changing the region settings for it to take effect.

Figure 146 VoIP > Phone > Region



Each field is described in the following table.

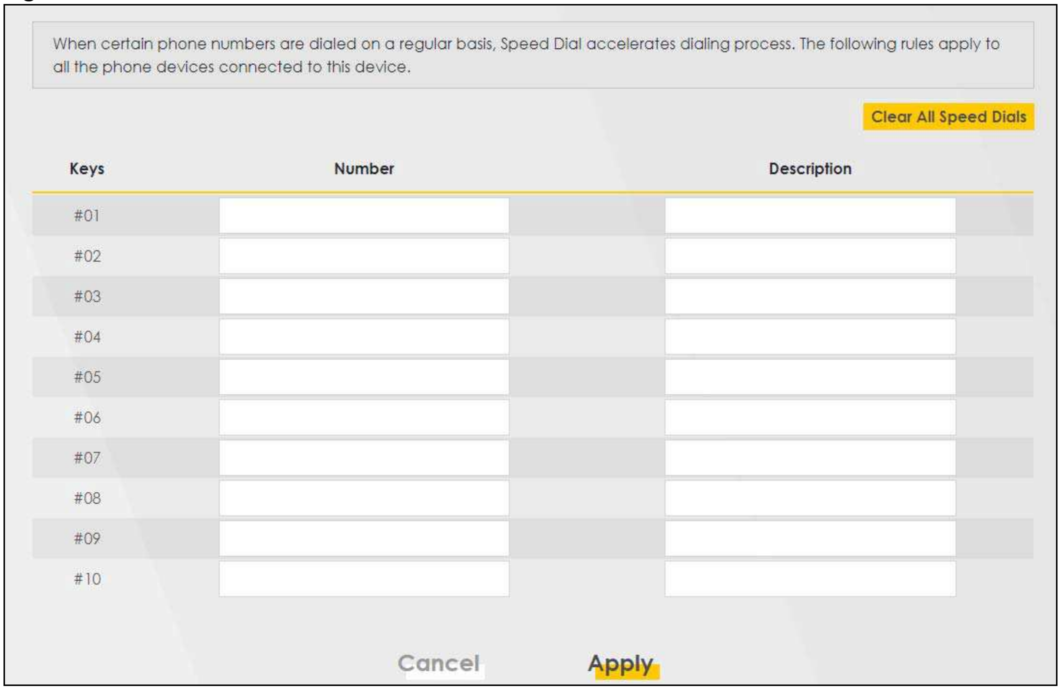
Table 102 VoIP > Phone > Region

LABEL	DESCRIPTION
Region Setting	Select the place in which the Zyxel Device is located.
Call Service Mode	Select the mode for supplementary phone services (call hold, call waiting, call transfer and three-way conference calls) that your VoIP service provider supports. **Europe Type** - use supplementary phone services in European mode **USA Type** - use supplementary phone services American mode You might have to subscribe to these services to use them. Contact your VoIP service provider.
Cancel	Click this to set every field in this screen to its last-saved value.
Apply	Click this to save your changes and to apply them to the Zyxel Device.

21.7 The Call Rule Screen

Use this screen to add, edit, or remove speed-dial numbers for outgoing calls. Speed dial provides shortcuts for dialing frequently-used (VoIP) phone numbers. You also have to create speed-dial entries if you want to call SIP addresses that contain letters. Once you have configured a speed dial rule, you can use a shortcut (the speed dial number, #01 for example) on your phone's keypad to call the phone number.

Figure 147 VoIP > Call Rule



Each field is described in the following table.

Table 103 VoIP > Call Rule

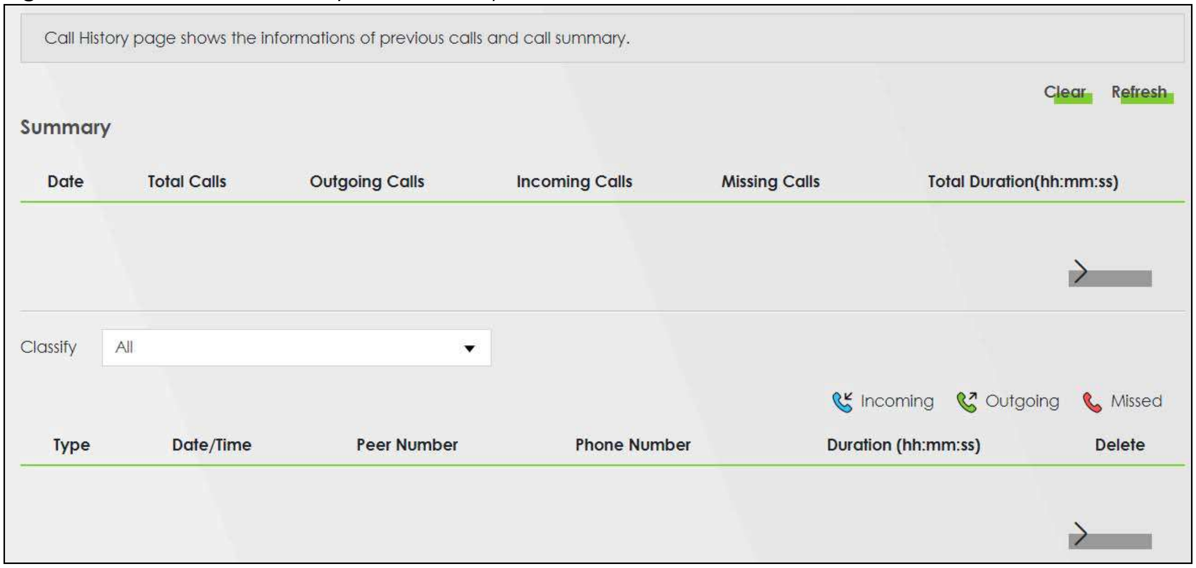
LABEL	DESCRIPTION
Clear All Speed Dials	Click this to erase all the speed-dial entries on this screen.
Keys	This field displays the speed-dial number you should dial to use this entry.
Number	Enter the SIP address you want the Zyxel Device to call when you dial the speed-dial number.
Description	Enter a name to identify the party you call when you dial the speed-dial number. You can use up to 127 printable ASCII characters.
Cancel	Click this to set every field in this screen to its last-saved value.
Apply	Click this to save your changes and to apply them to the Zyxel Device.

21.8 The Call History Screen

The Zyxel Device logs calls from or to your SIP addresses. This screen allows you to view the summary of received, dialed and missed calls and a call history list. You can also see detailed information for each outgoing call you made or each incoming call from someone calling you. The Zyxel Device stores up to 300 incoming call logs and 300 outgoing call logs. If the number of entries exceed the maximum value, the earliest log of that type will be deleted.

Click **VoIP > Call History > Call History**. The following screen displays.

Figure 148 VoIP > Call History > Call History



Each field is described in the following table.

Table 104 VoIP > Call History > Call History

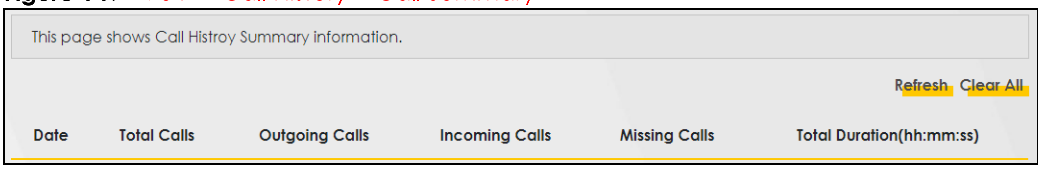
LABEL	DESCRIPTION
Clear ~~List~~	Click this button to remove all entries from the call history list.
Refresh	Click this button to renew the call history list.
~~Export~~	~~Click **Export** to download a call history list.~~
Date	This is the date when the calls were made.
Total Calls	This displays the total number of calls from or to your SIP addresses that day.
Outgoing Calls	This displays how many calls originated from you that day.
Incoming Calls	This displays how many calls you received that day.
Missing Calls	This displays how many incoming calls were not answered that day.
Total Duration (hh:mm:ss)	This displays how long all calls lasted that day.
Classify	Select the type of the calls. The call types are: **All**, **Incoming**, **Outgoing** and **Missed**.
Type	This displays the type of the calls.
Date/Time	This displays the date and time when the calls were made.
~~Name~~	~~This displays the SIP account name you called.~~
Peer Number	This displays the SIP address that called you or you called.
Phone Number	This displays the SIP address you used to make outgoing calls or receive calls.
~~Phone Device~~	~~This field displays the name of a phone port on the Zyxel Device.~~
~~Outgoing Number~~	~~This displays how many calls originated from you that day.~~
Duration (hh:mm:ss)	This displays how long the ~~current~~ call ~~has~~ lasted.
~~Modify~~Delete	Click the **Delete** icon to remove the call history.

21.9 ~~The Call Summary Screen~~

~~The Zyxel Device logs calls from or to your SIP addresses. This screen allows you to view the summary of received, dialed and missed calls.~~

~~Click **VoIP > Call History > Call Summary**. The following screen displays.~~

Figure 149 ~~VoIP > Call History > Call Summary~~



~~Each field is described in the following table.~~

Table 105 ~~VoIP > Call History > Call Summary~~

~~**LABEL**~~	~~**DESCRIPTION**~~
~~Refresh~~	~~Click this button to renew the call history list.~~
~~Clear All~~	~~Click this button to remove all entries from the call history list.~~
~~Date~~	~~This is the date when the calls were made.~~
~~Total Calls~~	~~This displays the total number of calls from or to your SIP addresses that day.~~
~~Outgoing Calls~~	~~This displays how many calls originated from you that day.~~
~~Incoming Calls~~	~~This displays how many calls you received that day.~~
~~Missing Calls~~	~~This displays how many incoming calls were not answered that day.~~
~~Total Duration (hh:mm:ss)~~	~~This displays how long all calls lasted that day.~~

21.10 Technical Reference

This section contains background material relevant to the **VoIP** screens.

VoIP

VoIP is the sending of voice signals over Internet Protocol. This allows you to make phone calls and send faxes over the Internet at a fraction of the cost of using the traditional circuit-switched telephone network. You can also use servers to run telephone service applications like PBX services and voice mail. Internet Telephony Service Provider (ITSP) companies provide VoIP service.

Circuit-switched telephone networks require 64 kilobits per second (Kbps) in each direction to handle a telephone call. VoIP can use advanced voice coding techniques with compression to reduce the required bandwidth.

SIP

The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol that handles the setting up, altering and tearing down of voice and multimedia sessions over the Internet.

SIP signaling is separate from the media for which it handles sessions. The media that is exchanged during the session can use a different path from that of the signaling. SIP handles telephone calls and can interface with traditional circuit-switched telephone networks.

SIP Identities

A SIP account uses an identity (sometimes referred to as a SIP address). A complete SIP identity is called a SIP URI (Uniform Resource Identifier). A SIP account's URI identifies the SIP account in a way similar to the way an email address identifies an email account. The format of a SIP identity is SIP-Number@SIP-Service-Domain.

SIP Number

The SIP number is the part of the SIP URI that comes before the “@” symbol. A SIP number can use letters like in an email address (johndoe@your-ITSP.com for example) or numbers like a telephone number (1122334455@VoIP-provider.com for example).

SIP Service Domain

The SIP service domain of the VoIP service provider is the domain name in a SIP URI. For example, if the SIP address is 1122334455@VoIP-provider.com, then “VoIP-provider.com” is the SIP service domain.

SIP Registration

Each Zyxel Device is an individual SIP User Agent (UA). To provide voice service, it has a public IP address for SIP and RTP protocols to communicate with other servers.

A SIP user agent has to register with the SIP registrar and must provide information about the users it represents, as well as its current IP address (for the routing of incoming SIP requests). After successful registration, the SIP server knows that the users (identified by their dedicated SIP URIs) are represented by the UA, and knows the IP address to which the SIP requests and responses should be sent.

Registration is initiated by the User Agent Client (UAC) running in the VoIP gateway (the Zyxel Device). The gateway must be configured with information letting it know where to send the REGISTER message, as well as the relevant user and authorization data.

A SIP registration has a limited lifespan. The User Agent Client must renew its registration within this lifespan. If it does not do so, the registration data will be deleted from the SIP registrar's database and the connection broken.

The Zyxel Device attempts to register all enabled subscriber ports when it is switched on. When you enable a subscriber port that was previously disabled, the Zyxel Device attempts to register the port immediately.

Authorization Requirements

SIP registrations (and subsequent SIP requests) require a username and password for authorization. These credentials are validated via a challenge / response system using the HTTP digest mechanism (as detailed in RFC 3261, "SIP: Session Initiation Protocol").

SIP Servers

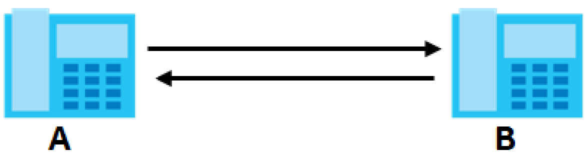
SIP is a client-server protocol. A SIP client is an application program or device that sends SIP requests. A SIP server responds to the SIP requests.

When you use SIP to make a VoIP call, it originates at a client and terminates at a server. A SIP client could be a computer or a SIP phone. One device can act as both a SIP client and a SIP server.

SIP User Agent

A SIP user agent can make and receive VoIP telephone calls. This means that SIP can be used for peer-to-peer communications even though it is a client-server protocol. In the following figure, either **A** or **B** can act as a SIP user agent client to initiate a call. **A** and **B** can also both act as a SIP SIP user agent to receive the call.

Figure 150 SIP User Agent



SIP Proxy Server

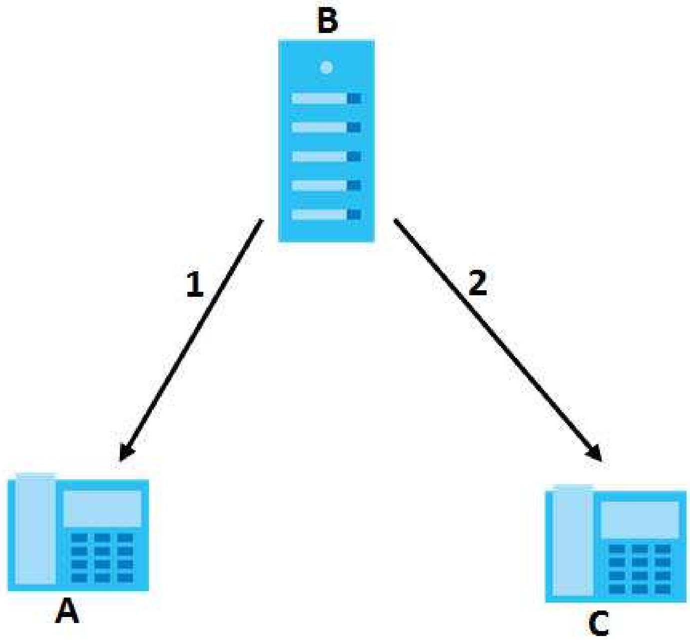
A SIP proxy server receives requests from clients and forwards them to another server.

In the following example, you want to use client device **A** to call someone who is using client device C.

1 The client device (**A** in the figure) sends a call invitation to the SIP proxy server (**B**).

2 The SIP proxy server forwards the call invitation to **C**.

Figure 151 SIP Proxy Server



SIP Redirect Server

A SIP redirect server accepts SIP requests, translates the destination address to an IP address and sends the translated IP address back to the device that sent the request. Then the client device that originally sent the request can send requests to the IP address that it received back from the redirect server. Redirect servers do not initiate SIP requests.

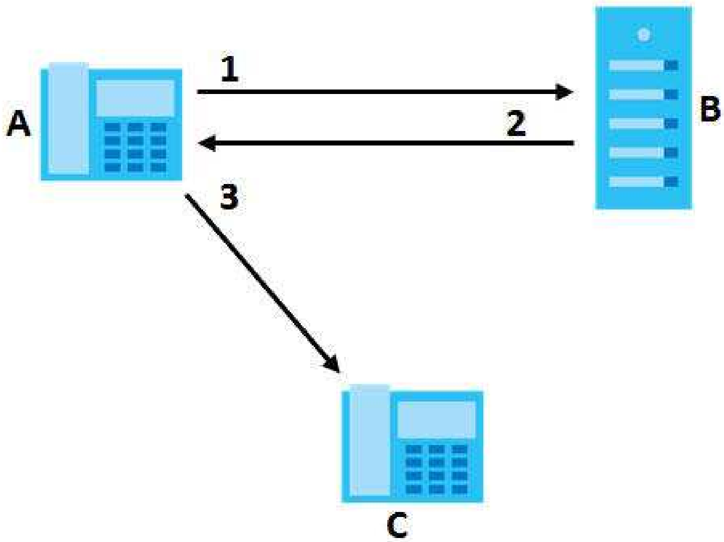
In the following example, you want to use client device **A** to call someone who is using client device **C**.

1 Client device **A** sends a call invitation for **C** to the SIP redirect server (**B**).

2 The SIP redirect server sends the invitation back to **A** with **C**'s IP address (or domain name).

3 Client device **A** then sends the call invitation to client device **C**.

Figure 152 SIP Redirect Server



SIP Register Server

A SIP register server maintains a database of SIP identity-to-IP address (or domain name) mapping. The register server checks your user name and password when you register.

RTP

When you make a VoIP call using SIP, the RTP (Real time Transport Protocol) is used to handle voice data transfer. See RFC 1889 for details on RTP.

Pulse Code Modulation

Pulse Code Modulation (PCM) measures analog signal amplitudes at regular time intervals and converts them into bits.

SIP Call Progression

The following figure displays the basic steps in the setup and tear down of a SIP call. A calls B.

Table 106 SIP Call Progression

A		B
1. INVITE	→	
	←	2. Ringing
	←	3. OK
4. ACK	→	
	5.Dialogue (voice traffic)	
6. BYE	→	
	←	7. OK

1 **A** sends a SIP INVITE request to **B**. This message is an invitation for **B** to participate in a SIP telephone call.

2 **B** sends a response indicating that the telephone is ringing.

3 **B** sends an OK response after the call is answered.

4 **A** then sends an ACK message to acknowledge that **B** has answered the call.

5 Now **A** and **B** exchange voice media (talk).

6 After talking, **A** hangs up and sends a BYE request.

7 **B** replies with an OK response confirming receipt of the BYE request and the call is terminated.

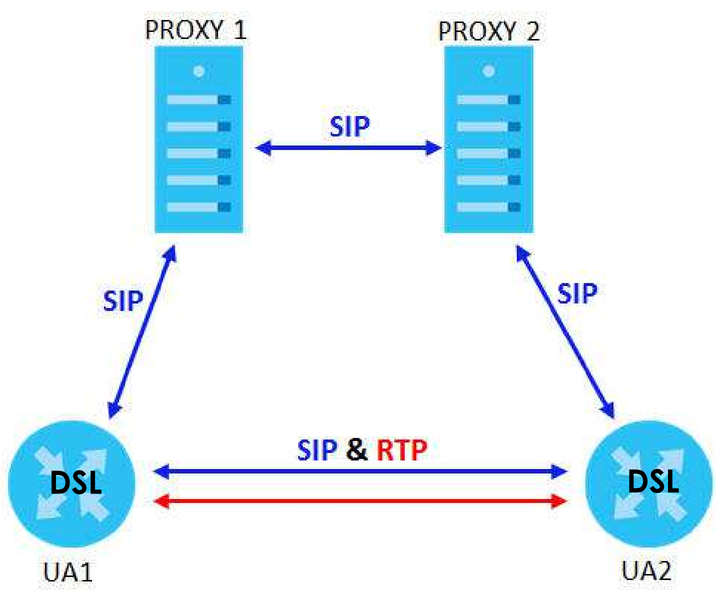
SIP Call Progression Through Proxy Servers

Usually, the SIP UAC sets up a phone call by sending a request to the SIP proxy server. Then, the proxy server looks up the destination to which the call should be forwarded (according to the URI requested by the SIP UAC). The request may be forwarded to more than one proxy server before arriving at its destination.

The response to the request goes to all the proxy servers through which the request passed, in reverse sequence. Once the session is set up, session traffic is sent between the UAs directly, bypassing all the proxy servers in between.

The following figure shows the SIP and session traffic flow between the user agents (**UA 1** and **UA 2**) and the proxy servers (this example shows two proxy servers, **PROXY 1** and **PROXY 2**).

Figure 153 SIP Call Through Proxy Servers



The following table shows the SIP call progression.

Table 107   SIP Call Progression

UA 1		PROXY 1		PROXY 2		UA 2
Invite	→					
		Invite	→			
	←	100 Trying		Invite	→	
			←	100 Trying		
					←	180 Ringing
			←	180 Ringing		
	←	180 Ringing				
					←	200 OK
			←	200 OK		
	←	200 OK				
ACK	→					
RTP	↔					RTP
	←					BYE
200 OK	→					

1 **User Agent 1** sends a SIP INVITE request to **Proxy 1**. This message is an invitation to **User Agent 2** to participate in a SIP telephone call. **Proxy 1** sends a response indicating that it is trying to complete the request.

2 **Proxy 1** sends a SIP INVITE request to **Proxy 2**. **Proxy 2** sends a response indicating that it is trying to complete the request.

3 **Proxy 2** sends a SIP INVITE request to **User Agent 2**.

4 **User Agent 2** sends a response back to **Proxy 2** indicating that the phone is ringing. The response is relayed back to **User Agent 1** via **Proxy 1**.

5 **User Agent 2** sends an OK response to **Proxy 2** after the call is answered. This is also relayed back to **User Agent 1** via **Proxy 1**.

6 **User Agent 1** and **User Agent 2** exchange RTP packets containing voice data directly, without involving the proxies.

7 When **User Agent 2** hangs up, he sends a BYE request.

8 **User Agent 1** replies with an OK response confirming receipt of the BYE request, and the call is terminated.

Voice Coding

A codec (coder/decoder) codes analog voice signals into digital signals and decodes the digital signals back into analog voice signals. The Zyxel Device supports the following codecs.

- G.711 is a Pulse Code Modulation (PCM) waveform codec. PCM measures analog signal amplitudes at regular time intervals and converts them into digital samples. G.711 provides very good sound quality but requires 64 kbps of bandwidth.

- G.726 is an Adaptive Differential PCM (ADPCM) waveform codec that uses a lower bitrate than standard PCM conversion. ADPCM converts analog audio into digital signals based on the difference between each audio sample and a prediction based on previous samples. The more similar the audio sample is to the prediction, the less space needed to describe it. G.726 operates at 16, 24, 32 or 40 kbps.
- G.729 is an Analysis-by-Synthesis (AbS) hybrid waveform codec that uses a filter based on information about how the human vocal tract produces sounds. G.729 provides good sound quality and reduces the required bandwidth to 8 kbps.

Voice Activity Detection/Silence Suppression

Voice Activity Detection (VAD) detects whether or not speech is present. This lets the Zyxel Device reduce the bandwidth that a call uses by not transmitting "silent packets" when you are not speaking.

Comfort Noise Generation

When using VAD, the Zyxel Device generates comfort noise when the other party is not speaking. The comfort noise lets you know that the line is still connected as total silence could easily be mistaken for a lost connection.

Echo Cancellation

G.168 is an ITU-T standard for eliminating the echo caused by the sound of your voice reverberating in the telephone receiver while you talk.

MWI (Message Waiting Indication)

Enable Message Waiting Indication (MWI) enables your phone to give you a message–waiting (beeping) dial tone when you have a voice message(s). Your VoIP service provider must have a messaging system that sends message waiting status SIP packets as defined in RFC 3842.

Custom Tones (IVR)

IVR (Interactive Voice Response) is a feature that allows you to use your telephone to interact with the Zyxel Device. The Zyxel Device allows you to record custom tones for the **Early Media** and **Music On Hold** functions. The same recordings apply to both the caller ringing and on hold tones.

Table 108 Custom Tones Details

LABEL	DESCRIPTION
Total Time for All Tones	900 seconds for all custom tones combined
Maximum Time per Individual Tone	180 seconds
Total Number of Tones Recordable	5 You can record up to 5 different custom tones but the total time must be 900 seconds or less.

Recording Custom Tones

Use the following steps if you would like to create new tones or change your tones:

1 Pick up the phone and press "****" on your phone's keypad and wait for the message that says you are in the configuration menu.

2 Press a number from 1101~1105 on your phone followed by the "#" key.

3 Play your desired music or voice recording into the receiver's mouthpiece. Press the "#" key.

4 You can continue to add, listen to, or delete tones, or you can hang up the receiver when you are done.

Listening to Custom Tones

Do the following to listen to a custom tone:

1 Pick up the phone and press "****" on your phone's keypad and wait for the message that says you are in the configuration menu.

2 Press a number from 1201~1208 followed by the "#" key to listen to the tone.

3 You can continue to add, listen to, or delete tones, or you can hang up the receiver when you are done.

Deleting Custom Tones

Do the following to delete a custom tone:

1 Pick up the phone and press "****" on your phone's keypad and wait for the message that says you are in the configuration menu.

2 Press a number from 1301~1308 followed by the "#" key to delete the tone of your choice. Press 14 followed by the "#" key if you wish to clear all your custom tones.

You can continue to add, listen to, or delete tones, or you can hang up the receiver when you are done.

21.10.1 Quality of Service (QoS)

Quality of Service (QoS) refers to both a network's ability to deliver data with minimum delay, and the networking methods used to provide bandwidth for real-time multimedia applications.

Type of Service (ToS)

Network traffic can be classified by setting the ToS (Type of Service) values at the data source (for example, at the Zyxel Device) so a server can decide the best method of delivery, that is the least cost, fastest route and so on.

DiffServ

DiffServ is a class of service (CoS) model that marks packets so that they receive specific per-hop treatment at DiffServ-compliant network devices along the route based on the application types and traffic flow. Packets are marked with DiffServ Code Points (DSCP) indicating the level of service desired.

This allows the intermediary DiffServ-compliant network devices to handle the packets differently depending on the code points without the need to negotiate paths or remember state information for every flow. In addition, applications do not have to request a particular service or give advanced notice of where the traffic is going.[3]

DSCP and Per-Hop Behavior

DiffServ defines a new DS (Differentiated Services) field to replace the Type of Service (TOS) field in the IP header. The DS field contains a 2-bit unused field and a 6-bit DSCP field which can define up to 64 service levels. The following figure illustrates the DS field.

DSCP is backward compatible with the three precedence bits in the ToS octet so that non-DiffServ compliant, ToS-enabled network device will not conflict with the DSCP mapping.

Figure 154 DiffServ: Differentiated Service Field

DSCP	Unused
(6-bit)	(2-bit)

The DSCP value determines the forwarding behavior, the PHB (Per-Hop Behavior), that each packet gets across the DiffServ network. Based on the marking rule, different kinds of traffic can be marked for different priorities of forwarding. Resources can then be allocated according to the DSCP values and the configured policies.

21.10.2 Phone Services Overview

Supplementary services such as call hold, call waiting, and call transfer. are generally available from your VoIP service provider. The Zyxel Device supports the following services:

- Call Return
- Call Hold
- Call Waiting
- Making a Second Call
- Call Transfer
- Call Forwarding
- Three-Way Conference
- Internal Calls
- Call Park and Pickup
- Do not Disturb
- IVR
- Call Completion
- CCBS
- Outgoing SIP

3. The Zyxel Device does not support DiffServ at the time of writing.

Note: To take full advantage of the supplementary phone services available through the Zyxel Device's phone ports, you may need to subscribe to the services from your VoIP service provider.

21.10.2.1 The Flash Key

Flashing means to press the hook for a short period of time (a few hundred milliseconds) before releasing it. On newer telephones, there should be a "flash" key (button) that generates the signal electronically. If the flash key is not available, you can tap (press and immediately release) the hook by hand to achieve the same effect. However, using the flash key is preferred since the timing is much more precise. With manual tapping, if the duration is too long, it may be interpreted as hanging up by the Zyxel Device.

You can invoke all the supplementary services by using the flash key.

21.10.2.2 Europe Type Supplementary Phone Services

This section describes how to use supplementary phone services with the **Europe Type Call Service Mode**. Commands for supplementary services are listed in the table below.

After pressing the flash key, if you do not issue the sub-command before the default sub-command timeout (2 seconds) expires or issue an invalid sub-command, the current operation will be aborted.

Table 109 European Flash Key Commands

COMMAND	SUB-COMMAND	DESCRIPTION
Flash		Put a current call on hold to place a second call. Switch back to the call (if there is no second call).
Flash	0	Drop the call presently on hold or reject an incoming call which is waiting for answer.
Flash	1	Disconnect the current phone connection and answer the incoming call or resume with caller presently on hold.
Flash	2	1. Switch back and forth between two calls. 2. Put a current call on hold to answer an incoming call. 3. Separate the current three-way conference call into two individual calls (one is on-line, the other is on hold).
Flash	3	Create three-way conference connection.
Flash	*98#	Transfer the call to another phone.

European Call Hold

Call hold allows you to put a call (**A**) on hold by pressing the flash key.

If you have another call, press the flash key and then “2” to switch back and forth between caller **A** and **B** by putting either one on hold.

Press the flash key and then “0” to disconnect the call presently on hold and keep the current call on line.

Press the flash key and then “1” to disconnect the current call and resume the call on hold.

If you hang up the phone but a caller is still on hold, there will be a remind ring.

European Call Waiting

This allows you to place a call on hold while you answer another incoming call on the same telephone (directory) number.

If there is a second call to a telephone number, you will hear a call waiting tone. Take one of the following actions.

- Reject the second call.
  Press the flash key and then press “0”.
- Disconnect the first call and answer the second call.
  Either press the flash key and press “1”, or just hang up the phone and then answer the phone after it rings.
- Put the first call on hold and answer the second call.
  Press the flash key and then “2”.

European Call Transfer

Do the following to transfer an incoming call (that you have answered) to another phone.

1 Press the flash key to put the caller on hold.

2 When you hear the dial tone, dial “*98#” followed by the number to which you want to transfer the call.

3 After you hear the ring signal or the second party answers it, hang up the phone.

European Three-Way Conference

Use the following steps to make three-way conference calls.

1 When you are on the phone talking to someone, press the flash key to put the caller on hold and get a dial tone.

2 Dial a phone number directly to make another call.

3 When the second call is answered, press the flash key and press “3” to create a three-way conversation.

4 Hang up the phone to drop the connection.

5 If you want to separate the activated three-way conference into two individual connections (one is on-line, the other is on hold), press the flash key and press “2”.

21.10.2.3 USA Type Supplementary Services

This section describes how to use supplementary phone services with the **USA Type Call Service Mode**. Commands for supplementary services are listed in the table below.

After pressing the flash key, if you do not issue the sub-command before the default sub-command timeout (2 seconds) expires or issue an invalid sub-command, the current operation will be aborted.

Table 110   USA Flash Key Commands

COMMAND	SUB-COMMAND	DESCRIPTION
Flash		Put a current call on hold to place a second call. After the second call is successful, press the flash key again to have a three-way conference call. Put a current call on hold to answer an incoming call.
Flash	*98#	Transfer the call to another phone.

USA Call Hold

Call hold allows you to put a call (**A**) on hold by pressing the flash key.

If you have another call, press the flash key to switch back and forth between caller **A** and **B** by putting either one on hold.

If you hang up the phone but a caller is still on hold, there will be a remind ring.

USA Call Waiting

This allows you to place a call on hold while you answer another incoming call on the same telephone (directory) number.

If there is a second call to your telephone number, you will hear a call waiting tone.

Press the flash key to put the first call on hold and answer the second call.

USA Call Transfer

Do the following to transfer an incoming call (that you have answered) to another phone.

1 Press the flash key to put the caller on hold.

2 When you hear the dial tone, dial "*98#" followed by the number to which you want to transfer the call.

3 After you hear the ring signal or the second party answers it, hang up the phone.

USA Three-Way Conference

Use the following steps to make three-way conference calls.

1 When you are on the phone talking to someone (party A), press the flash key to put the caller on hold and get a dial tone.

2 Dial a phone number directly to make another call (to party B).

3 When party B answers the second call, press the flash key to create a three-way conversation.

4 Hang up the phone to drop the connection.

5 If you want to separate the activated three-way conference into two individual connections (with party A on-line and party B on hold), press the flash key.

6 If you want to go back to the three-way conversation, press the flash key again.

7 If you want to separate the activated three-way conference into two individual connections again, press the flash key. This time the party B is on-line and party A is on hold.

21.10.2.4 Phone Functions Summary

The following table shows the key combinations you can enter on your phone's keypad to use certain features.

Table 111 Phone Functions Summary

ACTION	FUNCTION	DESCRIPTION
*98#	Call transfer	Transfer a call to another phone. See Section 21.10.2.2 on page 254 (Europe type) and Section 21.10.2.3 on page 255 (USA type).
*66#	Call return	Place a call to the last person who called you.
*95#	Enable Do Not Disturb	Use these to set your phone not to ring when someone calls you, or to turn this function off.
#95#	Disable Do Not Disturb	
*41#	Enable Call Waiting	Use these to allow you to put a call on hold when you are answering another, or to turn this function off.
#41#	Disable Call Waiting	
****	IVR	Use these to set up Interactive Voice Response (IVR). IVR allows you to record custom caller ringing tones (the sound a caller hears before you pick up the phone) and on hold tones (the sound someone hears when you put their call on hold).
####	Internal Call	Call the phone(s) connected to the Zyxel Device.
*82	One Shot Caller Display Call	Activate or deactivate caller ID for the next call only.
*67	One Shot Caller Hidden Call	

CHAPTER 22 Log

22.1 Log Overview

These screens allow you to determine the categories of events and/or alerts that the Zyxel Device logs and then display these logs or have the Zyxel Device send them to an administrator (through email) or to a syslog server.

22.1.1 What You Can Do in this Chapter

- Use the **System Log** screen to see the system logs (Section 22.2 on page 259).
- Use the **Security Log** screen to see the security-related logs for the categories that you select (Section 22.3 on page 260).

22.1.2 What You Need To Know

The following terms and concepts may help as you read this chapter.

Alerts and Logs

An alert is a type of log that warrants more serious attention. They include system errors, attacks (access control) and attempted access to blocked web sites. Some categories such as **System Errors** consist of both logs and alerts. You may differentiate them by their color in the **View Log** screen. Alerts display in red and logs display in black.

Syslog Overview

The syslog protocol allows devices to send event notification messages across an IP network to syslog servers that collect the event messages. A syslog-enabled device can generate a syslog message and send it to a syslog server.

Syslog is defined in RFC 3164. The RFC defines the packet format, content and system log related information of syslog messages. Each syslog message has a facility and severity level. The syslog facility identifies a file in the syslog server. Refer to the documentation of your syslog program for details. The following table describes the syslog severity levels.

Table 112 Syslog Severity Levels

CODE	SEVERITY
0	Emergency: The system is unusable.
1	Alert: Action must be taken immediately.
2	Critical: The system condition is critical.
3	Error: There is an error condition on the system.
4	Warning: There is a warning condition on the system.

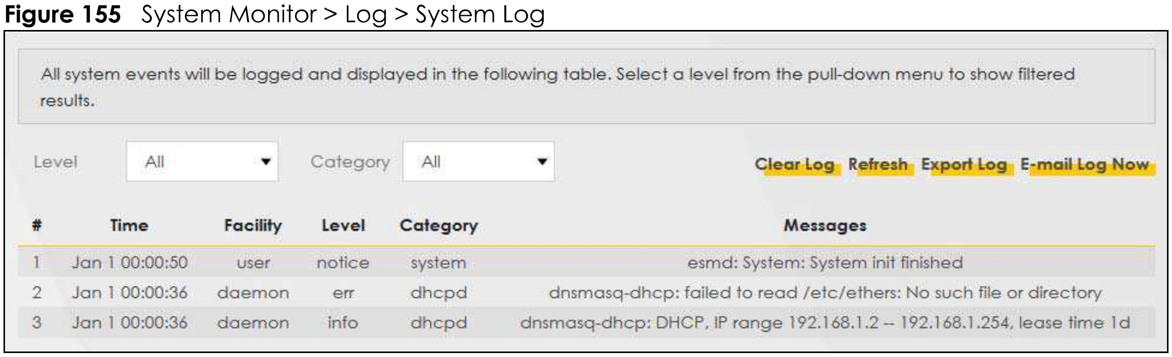
Table 112 Syslog Severity Levels

CODE	SEVERITY
5	Notice: There is a normal but significant condition on the system.
6	Informational: The syslog contains an informational message.
7	Debug: The message is intended for debug-level purposes.

22.2 System Log

Use the **System Log** screen to see the system logs. You can filter the entries by ~~clicking on the~~ ~~**Level**~~selecting a severity level and/or category~~**Category** drop-down list boxes~~. Click **System Monitor > Log > System Log** to open the **System Log** screen.

Figure 155 System Monitor > Log > System Log



The following table describes the fields in this screen.

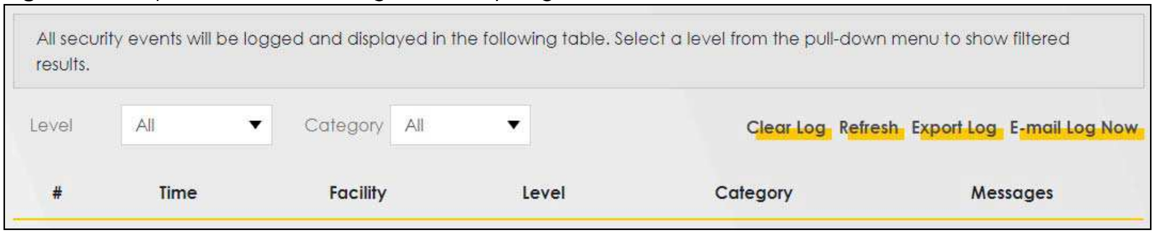
Table 113 System Monitor > Log > System Log

LABEL	DESCRIPTION
Level	Select a severity level from the drop-down list box. This filters search results according to the severity level you have selected. When you select a severity, the Zyxel Device searches through all logs of that severity or higher.
Category	Select the type of logs to display.
Clear Log	Click this to delete all the logs.
Refresh	Click this to renew the log screen.
Export Log	Click this to ~~export the selected log(s)~~save the current list of logs to your computer.
E-mail Log Now	Click this to send the log file(s) to the Email address you specify in the **Maintenance >** ~~**Logs Setting**~~**E-mail Notification** screen.
#	This field is a sequential value and is not associated with a specific entry.
Time	This field displays the time the log was recorded.
Facility	The log facility allows you to send logs to different files in the syslog server. Refer to the documentation of your syslog program for more details.
Level	This field displays the severity level of the log~~ that the device is to send to this syslog server~~.
Category	This field displays the type of the log.
Messages	This field states the reason for the log.

22.3 Security Log

Use the **Security Log** screen to see the security-related logs for the categories that you select. You can filter the entries by ~~clicking on the **Level** and/or **Category** drop-down list boxes~~selecting a severity level and/or category. Click **System Monitor > Log > Security Log** to open the following screen.

Figure 156 System Monitor > Log > Security Log



The following table describes the fields in this screen.

Table 114 System Monitor > Log > Security Log

LABEL	DESCRIPTION
Level	Select a severity level from the drop-down list box. This filters search results according to the severity level you have selected. When you select a severity, the Zyxel Device searches through all logs of that severity or higher.
Category	Select the type of logs to display.
Clear Log	Click this to delete all the logs.
Refresh	Click this to renew the log screen.
Export Log	Click this to ~~export the selected log(s)~~save the current list of logs to your computer.
Email Log Now	Click this to send the log file(s) to the Email address you specify in the **Maintenance >** ~~**Logs Setting**~~**E-mail Notification** screen.
#	This field is a sequential value and is not associated with a specific entry.
Time	This field displays the time the log was recorded.
Facility	The log facility allows you to send logs to different files in the syslog server. Refer to the documentation of your syslog program for more details.
Level	This field displays the severity level of the log~~ that the device is to send to this syslog server~~.
Category	This field displays the type of the log.
Messages	This field states the reason for the log.

CHAPTER 23
Traffic Status

23.1 Traffic Status Overview

Use the **Traffic Status** screens to look at the network traffic status and statistics of the WAN/LAN interfaces and NAT.

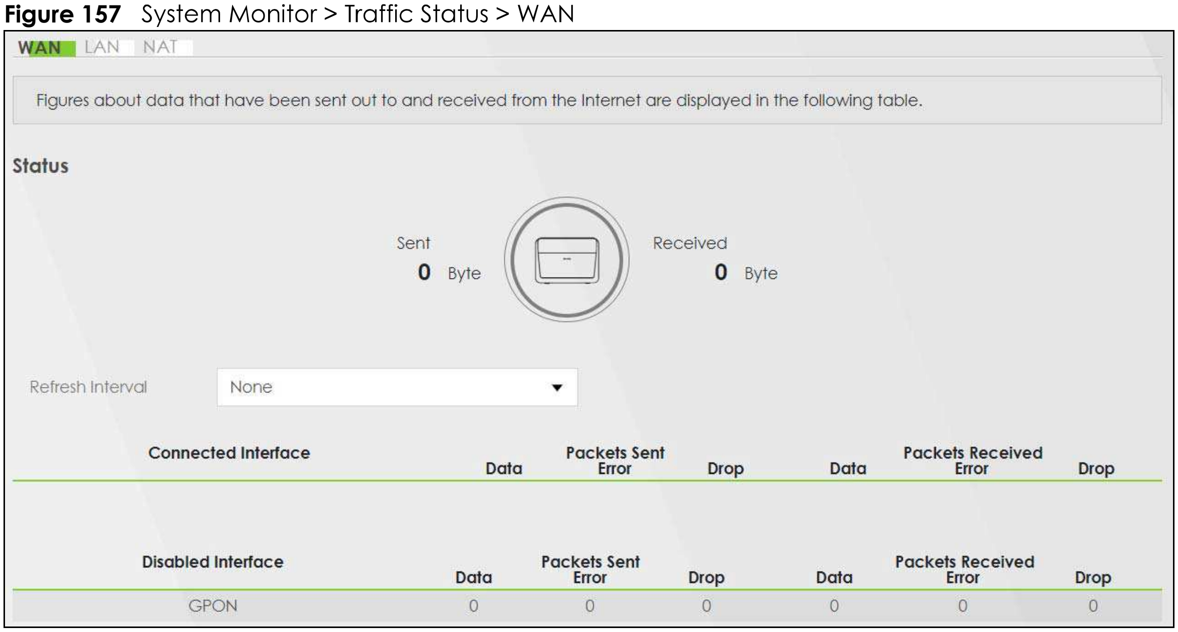
23.1.1 What You Can Do in this Chapter

- Use the **WAN** screen to view the WAN traffic statistics (Section 23.2 on page 261).
- Use the **LAN** screen to view the LAN traffic statistics (Section 23.3 on page 262).
- Use the **NAT** screen to view the NAT status of the Zyxel Device's client(s) (Section 23.4 on page 263).

23.2 WAN Status

Click **System Monitor > Traffic Status** to open the **WAN** screen. The figures in this screen show the total number of bytes received and sent through the Zyxel Device's WAN interface(s). Packet statistics for each WAN interface are listed in the tables below.

Figure 157 System Monitor > Traffic Status > WAN



The following table describes the fields in this screen.

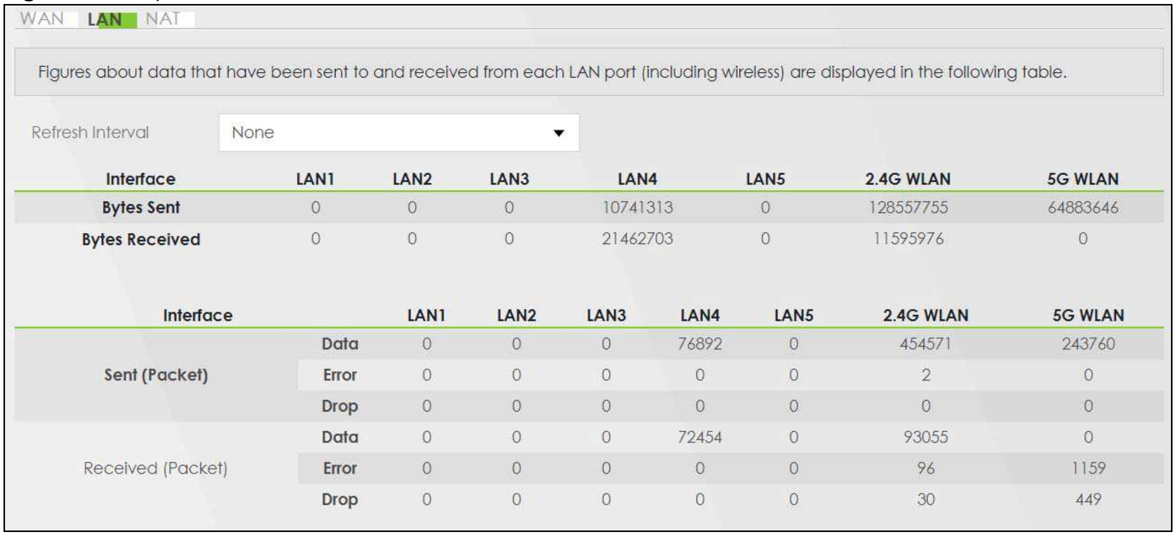
Table 115 System Monitor > Traffic Status > WAN

LABEL	DESCRIPTION
Refresh Interval	Select how often you want the Zyxel Device to update this screen.
Connected Interface	This shows the name of the WAN interface that is currently connected.
Packets Sent	
Data	This indicates the number of transmitted packets on this interface.
Error	This indicates the number of frames with errors transmitted on this interface.
Drop	This indicates the number of outgoing packets dropped on this interface.
Packets Received	
Data	This indicates the number of received packets on this interface.
Error	This indicates the number of frames with errors received on this interface.
Drop	This indicates the number of received packets dropped on this interface.
Disabled Interface	This shows the name of the WAN interface that is currently disabled.
Packets Sent	
Data	This indicates the number of transmitted packets on this interface.
Error	This indicates the number of frames with errors transmitted on this interface.
Drop	This indicates the number of outgoing packets dropped on this interface.
Packets Received	
Data	This indicates the number of received packets on this interface.
Error	This indicates the number of frames with errors received on this interface.
Drop	This indicates the number of received packets dropped on this interface.

23.3 LAN Status

Click **System Monitor > Traffic Status > LAN** to open the following screen. ~~The figures in this screen show the number of bytes received and sent from each LAN port and wireless network~~This screen allows you to view packet statistics for each LAN or WLAN interface on the Zyxel Device.

Figure 158 System Monitor > Traffic Status > LAN



The following table describes the fields in this screen.

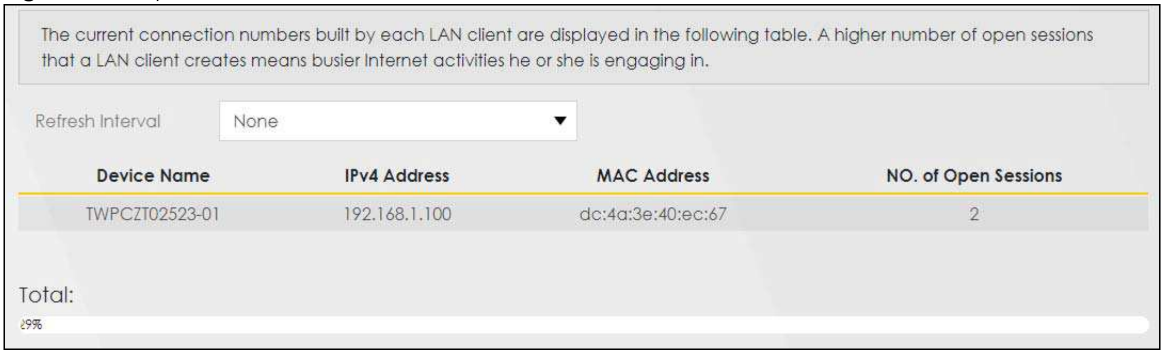
Table 116 System Monitor > Traffic Status > LAN

LABEL	DESCRIPTION
Refresh Interval	Select how often you want the Zyxel Device to update this screen.
Interface	This shows the LAN or wireless LAN interface on the Zyxel Device.
Bytes Sent	This indicates the number of bytes transmitted on this interface.
Bytes Received	This indicates the number of bytes received on this interface.
Interface	This shows the LAN or wireless LAN interfaces on the Zyxel Device.
Sent (Packets)	
Data	This indicates the number of transmitted packets on this interface.
Error	This indicates the number of frames with errors transmitted on this interface.
Drop	This indicates the number of outgoing packets dropped on this interface.
Received (Packets)	
Data	This indicates the number of received packets on this interface.
Error	This indicates the number of frames with errors received on this interface.
Drop	This indicates the number of received packets dropped on this interface.

23.4 NAT Status

Click **System Monitor > Traffic Status > NAT** to open the following screen. ~~The figures in this screen show the NAT session statistics for hosts currently connected to the Zyxel Device. A number of open sessions means there's a lot of traffic going through a host.~~This screen lists the devices that have received an IP address from the Zyxel Device LAN or WLAN interface(s) and have ever established a session with the Zyxel Device.

Figure 159 System Monitor > Traffic Status > NAT



The following table describes the fields in this screen.

Table 117 System Monitor > Traffic Status > NAT

LABEL	DESCRIPTION
Refresh Interval	Select how often you want the Zyxel Device to update this screen.
Device Name	This displays the name of the connected host.
IPv4 Address	This displays the IP address of the connected host.
MAC Address	This displays the MAC address of the connected host.
No. of Open Session	This displays the number of NAT sessions currently opened for the connected host.
Total	This displays what percentage of NAT sessions the Zyxel Device can support is currently being used by all connected hosts. ~~You can also see the number of active NAT sessions and the maximum number of NAT sessions the Zyxel Device can support.~~