

OpenScape Office V3

Tutorial

SIP Endpoint Configuration – Grandstream-Phones

Version 1.0

1 Grandstream phones

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1.1 Grandstream GXP280



For information see the Grandstream homepage:

http://www.grandstream.com/products/gxp_series/gxp280/gxp280.html

Used Endpoint:

Produkt-Modell: GXP280 (HW0.3B)

Software Version: Programm-- 1.2.3.5 Bootloader-- 1.1.6.8

1.1.1 Basic Configuration

Default Administrator password: "admin"

Basic Settings

If no DHCP is used, enter the IP network configuration parameters as used in your network:

Grandstream Device Configuration				
STATUS	BASIC SETTINGS		ADVANCED SETTINGS	ACCOUNT
End User Password: <input type="text"/> (purposely not displayed for security protection)				
IP Address: <input type="radio"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)				
PPPoE account ID:		<input type="text"/>		
PPPoE password:		<input type="text"/>		
Host name (Option 12):		<input type="text"/>		
Domain name (Option 15):		<input type="text"/>		
Vendor Class ID (Option 60):		<input type="text" value="Grandstream GXP280"/>		
Preferred DNS server: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>				
<input checked="" type="radio"/> statically configured as:				
IP Address:	<input type="text" value="192"/>	<input type="text" value="168"/>	<input type="text" value="138"/>	<input type="text" value="193"/>
Subnet Mask:	<input type="text" value="255"/>	<input type="text" value="255"/>	<input type="text" value="255"/>	<input type="text" value="0"/>
Gateway:	<input type="text" value="192"/>	<input type="text" value="168"/>	<input type="text" value="138"/>	<input type="text" value="249"/>
DNS Server 1:	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>
DNS Server 2:	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>

To get the correct time display set

- Daylight Saving Time
- Time Display Format
- Date Display Format
- Display Clock instead of Date

according to your needs:

Time Zone:	GMT+1:00 (Paris, Amsterdam, Berlin, Rome, Vienna, Madrid, Warsaw, Brussels) ▼
	Allow DHCP Option 2 to override Time Zone setting: <input checked="" type="radio"/> No <input type="radio"/> Yes
Daylight Savings Time:	<input type="radio"/> No <input checked="" type="radio"/> Yes
	Optional Rule: 3.2.7.2.0.11.1.7.2.0.60
Time Display Format:	<input type="radio"/> 12 HOUR <input checked="" type="radio"/> 24 HOUR
	<input type="radio"/> Year-Month-Day
Date Display Format:	<input type="radio"/> Month-Day-Year
	<input checked="" type="radio"/> Day-Month-Year
Display Clock instead of Date:	<input type="radio"/> No <input checked="" type="radio"/> Yes

Advanced settings:

Enter the IP-Address of your OpenScope Office as NTP server here:

NTP Server:	192.168.138.72 (URI or IP address)
	Allow DHCP Option 42 to override NTP server: <input checked="" type="radio"/> No <input type="radio"/> Yes

Advanced settings:

The following settings should be left in default

Grandstream Device Configuration			
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	ACCOUNT
	Admin Password: <input type="text"/> (purposely not displayed for security protection)		
	G723 rate: <input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate		
	iLBC frame size: <input checked="" type="radio"/> 20ms <input type="radio"/> 30ms		
	iLBC payload type: <input type="text" value="97"/> (between 96 and 127, default is 97)		
	Silence Suppression: <input checked="" type="radio"/> No <input type="radio"/> Yes		
	Voice Frames per TX: <input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)		
	Layer 3 QoS: <input type="text" value="48"/> (Diff-Serv or Precedence value)		
	Layer 2 QoS: 802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)		
	Data VLAN Tag: 1: <input type="text" value="0"/> 2: <input type="text" value="0"/> 3: <input type="text" value="0"/> (can't use the same non-zero value as 802.1Q tag)		
	No Key Entry Timeout: <input type="text" value="4"/> (in seconds, default is 4 seconds)		
	Use # as Dial Key: <input type="radio"/> No <input checked="" type="radio"/> Yes		
	local RTP port: <input type="text" value="5004"/> (1024-65400, default 5004, must be even)		
	Use random port: <input checked="" type="radio"/> No <input type="radio"/> Yes		
	keep-alive interval: <input type="text" value="20"/> (in seconds, default 20 seconds)		
	Use NAT IP: <input type="text"/> (if specified, this will be used in SIP/SDP message)		
	STUN server: <input type="text"/> (URI or IP:port)		

If you have to update the phone SW, provide the address of your TFTP server here. In case you want to have automatic updates enabled e.g. during reboot, set the flags accordingly.

Firmware Upgrade and Provisioning:	Upgrade Via: <input checked="" type="radio"/> TFTP <input type="radio"/> HTTP
	Firmware Server Path: <input type="text" value="192.168.138.12"/>
	Config Server Path: <input type="text"/>
	Firmware File Prefix: <input type="text"/>
	Firmware File Postfix: <input type="text"/>
	Config File Prefix: <input type="text"/>
	Config File Postfix: <input type="text"/>
	Allow DHCP Option43 and Option 66 to override server: <input checked="" type="radio"/> No <input type="radio"/> Yes
	Automatic Upgrade: <input checked="" type="radio"/> No <input type="radio"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days)
	<input type="radio"/> Always Check for New Firmware
	<input type="radio"/> Check New Firmware only when F/W pre/suffix changes
	<input checked="" type="radio"/> Always Skip the Firmware Check
	Authenticate Conf File: <input checked="" type="radio"/> No <input type="radio"/> Yes (cfg file would be authenticated before acceptance if set to Yes)
Phonebook XML Download:	Enable Phonebook XML Download: <input checked="" type="radio"/> No <input type="radio"/> YES, HTTP <input type="radio"/> YES, TFTP
	Phonebook XML Server Path: <input type="text"/>
	Phonebook Download Interval: <input type="text" value="0"/> (0-720, in minutes)
	Remove Manually-edited entries on Download: <input checked="" type="radio"/> No <input type="radio"/> Yes
LDAP Directory:	LDAP Script Server Path: <input type="text"/>
Offhook Auto Dial:	<input type="text"/> (User ID/extension to dial automatically when offhook, max length 35)
DTMF Payload Type:	<input checked="" type="text" value="101"/>
Onhook Threshold:	<input type="text" value="800 ms"/>

The following entries can be left in default (North American tones). If local tones are required this has to be changed.

Distinctive Ring Tone:	Custom ring tone 1, used if incoming caller ID is <input type="text"/>
	Custom ring tone 2, used if incoming caller ID is <input type="text"/>
	Custom ring tone 3, used if incoming caller ID is <input type="text"/>
System Ring Tone:	<input type="text" value="f1=440,f2=480,c=200/400;"/>
Call Progress Tones:	Dial Tone <input type="text" value="f1=350,f2=440;"/>
	Message Waiting <input type="text" value="f1=350,f2=440,c=10/10;"/>
	Ring Back Tone <input type="text" value="f1=440,f2=480,c=200/400;"/>
	Call-Waiting Tone <input type="text" value="f1=440,f2=440,c=25/525;"/>
	Busy Tone <input type="text" value="f1=480,f2=620,c=50/50;"/>
	Reorder Tone <input type="text" value="f1=480,f2=620,c=25/25;"/>
	Syntax: f1=val, f2=val [, c=on1/off1[-on2/off2[-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms)

Disable not supported features, this will hide this features on the UI

Disable Direct IP Calls:	<input type="radio"/> No <input checked="" type="radio"/> Yes
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If you want to use a different language, you have to select “secondary Language” and provide the corresponding language file via TFTP. See downloadchapter

Display Language:	<input type="radio"/> English <input type="radio"/> Chinese <input checked="" type="radio"/> Secondary Language <input type="text" value="ger"/> (Language File postfix)
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>	
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Registration and Basic Telephony

Account settings:

Phone Value	configured in OpenScope Office:
SIP-Server:	IP-Address of OpenScope Office
	configured in OpenScope Office: Telephones / Subscribers-> IP Telephones -> Edit
SIP User ID:	Call number
Authenticate Password:	Password
Authenticate ID :	Client-SIP User ID
Name	Optional, Phone name can only be seen in the network traces, OpenScope Office uses the name configured in system

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
Account Name: <input type="text" value=""/> (e.g., MyCompany) SIP Server: <input type="text" value="192.168.138.29"/> (e.g., sip.mycompany.com, or IP address) Outbound Proxy: <input type="text" value=""/> (e.g., proxy.myprovider.com, or IP address) SIP User ID: <input type="text" value="3760"/> (the user part of an SIP address) Authenticate ID: <input type="text" value="SIP-3760"/> (can be same or different from SIP UserID) Authenticate Password: <input type="password" value="*****"/> (not displayed for security protection) Name: <input type="text" value="GXP280-3760"/> (optional, e.g., John Doe) Use DNS SRV: <input checked="" type="radio"/> No <input type="radio"/> Yes User ID is phone number: <input type="radio"/> No <input checked="" type="radio"/> Yes SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes	

Send DTMF: disable in-audio, enable via RTP (RFC2833)

Send DTMF:	<input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
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Adjust the codec settings if needed:

Preferred Vocoder: (in listed order)	choice 1:	<input type="text" value="PCMU"/>	choice 5:	<input type="text" value="G.726-32"/>
	choice 2:	<input type="text" value="PCMA"/>	choice 6:	<input type="text" value="iLBC"/>
	choice 3:	<input type="text" value="G.729A/B"/>	choice 7:	<input type="text" value="G.723.1"/>
	choice 4:	<input type="text" value="G.722 (wide band)"/>	choice 8:	<input type="text" value="GSM"/>

Special deployment

Change Language:

The GXP280 comes with two different languages (English,Chinese)

If you want to have a different language it has to be downloaded via TFTP.

A language pack (GXP_Language_Pack.zip) is available at the Grandstream download site.

<http://www.grandstream.com/firmware.html#note8>


This language pack has the compiled file which is read to be used for GXP series. Each zip file has only one particular language in it.

How to use:

1. Open the zip file
2. Open the desired language zip file
3. Copy the gxp.lpf to the TFTP server path and rename it with a postfix e.g. gxp_ger.lpf
4. Check that your TFTP Server is running.
5. Access the advance setting of the Web UI, select Secondary Language and enter postfix e.g. "ger" without the "_"
6. Save and reboot the phone

1.1.2 Hold/Retrieve/Alternate

Pressing the "Flash" key will put a call on HOLD or retrieved it from HOLD. A consultation call can be established when a call is held. Toggle/alternate can be invoked by pressing the flash key during consultation.

	HOLD and all features based on HOLD will be disabled when "Send Flash Event" is set to Yes.
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Send Flash Event: <input checked="" type="radio"/> No <input type="radio"/> Yes

1.1.3 Transfer

Attended -, Semi-Attended- and Blind Transfer is supported.

Semi Attended Transfer Mode MUST be set to "Send REFER with early dialog". If set to RFC5589 (default) the transferor will remain busy until the transfer target accepts the call.

Semi-attended Transfer Mode: <input type="radio"/> RFC5589 <input checked="" type="radio"/> Send REFER with early dialog
--

Transfer can be disabled:.

Disable Transfer: <input checked="" type="radio"/> No <input type="radio"/> Yes

1.1.4 CLIP/CLIR/CNIP - Name and Number presentation

The phone can display names (default) or the call number

Display CID instead of Name:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
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Enable CLIR if required, by setting

Send Anonymous Yes

Anonymous Method Use Privacy Header

Send Anonymous:	<input checked="" type="radio"/> No	<input type="radio"/> Yes (caller ID will be blocked if set to Yes)
Anonymous Method:	<input checked="" type="radio"/> Use From Header	<input type="radio"/> Use Privacy Header

1.1.5 Call Waiting / Call offer

Call waiting is enabled by default in GXP280 but has to be enabled in OpenScape Office WBM. As this is a station oriented parameter there is no need to configure it in the phone. Nevertheless two parameters are provided:

Disable Call-Waiting:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Disable Call-Waiting Tone:	<input checked="" type="radio"/> No	<input type="radio"/> Yes

1.1.6 Call Forwarding

The endpoint offers

- CFU Always Call Forwarding unconditional

CF has to be activated/deactivated on the phone via a predefined soft key

1.1.7 Message Waiting

For this feature the “Account Settings”

- Subscribe for MWI
- Voice Mail UserID: Access number of VM

have to be configured.

SUBSCRIBE for MWI:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
SUBSCRIBE for Registration Event:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Proxy-Require:	<input type="text"/>	
Voice Mail UserID:	<input type="text" value="71"/>	(UserID for voice mail system)

A waiting message is signaled by a red light on top of the phone.

1.1.8 Distinctive Ringing

Not supported by GXP280

1.1.9 Local phone features

- DND – Do Not Disturb

The MUTE key can be used to invoke DND.

The feature can be deactivated by administration

Disable DND Button: ☒ No ☐ Yes (MUTE/DEL button pressing will have no effect if set to Yes)

- Conference

GXP280 offers a local 3 party conference. Active and held call can be connected to a 3 way conference by pressing the CONF key.

The feature can be deactivated by administration

Disable Conference: ☒ No ☐ Yes

1.1.10 Known limitations and restrictions

1.2 Grandstream GXV3140



For information see the Grandstream homepage:

http://www.grandstream.com/products/gxv_series_phone/gxv3140/gxv3140.html

Used Endpoint:

A screenshot of the GXV3140 Multimedia Phone Administration Interface. The interface has a blue header with the Grandstream logo and the title 'GXV3140 Multimedia Phone Administration Interface'. Below the header is a navigation bar with tabs: 'Status' (highlighted), 'Account 1', 'Account 2', 'Account 3', 'Advanced Settings', 'Maintenance', and 'Application Settings'. On the left side, there is a sidebar menu with 'Status' (highlighted), 'Account Status', 'Network Status', and 'System Info'. The main content area displays 'System Info' with a table of system details.

System Info	
Product Model	GXV3140
Hardware Revision	V0.4A
PN Code	9630001104A
Boot Version	1.0.3.2
Core Version	1.0.3.4
DSP Version	1.0.3.25
Base Version	1.0.3.16
Program Version	1.0.3.24
GUI-A Version	1.0.3.3
GUI-B Version	1.0.3.3
System Up Time	25 minutes, 48 seconds

Product highlights:

3 line multimedia phone with integrated video, multimedia player, Internet radio, IM client ...

1.2.1 Basic Configuration

Default Administrator login “admin”, password: “admin”

The phone supports up to 3 lines to make establish calls.



To allow features like consultation or conference at least two accounts have to be configured in the phone with identical configuration parameters.
EXCEPTION: Only for account 1 the flag SIP registration=yes is activated.

The screenshot shows the 'General Settings' page for 'Account 1' in the GXV3140 Multimedia Phone Administration Interface. The interface has a blue header with the Grandstream logo and a navigation bar with tabs: Status, Account 1 (selected), Account 2, Account 3, Advanced Settings, Maintenance, and Application Settings. On the left, there is a sidebar with a tree view for 'Account 1' containing: General Settings (selected), Network Settings, SIP Settings, Codec Settings, and Call Settings. The main content area is titled 'General Settings' and contains the following fields:

- Account Active: ☒ Yes
- Account Name:
- SIP Server:
- SIP User ID:
- Authenticate ID:
- Authenticate Password:
- Voice Mail UserID:
- Name:
- User ID is phone number: ☐ Yes

At the bottom of the form are 'Save' and 'Cancel' buttons.

For endpoints connected to the LAN NAT Traversal MUST be set to NO

The screenshot shows the 'Network Settings' page for 'Account 1' in the GXV3140 Multimedia Phone Administration Interface. The interface is similar to the previous one, with the 'Network Settings' tab selected in the sidebar. The main content area is titled 'Network Settings' and contains the following fields:

- Outbound Proxy:
- DNS Mode:
- NAT Traversal: (This field is highlighted with a red rectangle in the original image)
- Proxy-Require:

At the bottom of the form are 'Save' and 'Cancel' buttons.

Configure the Account SIP settings, SIP registration and SUBSCRIBE for MWI MUST be set only for Account 1 (primary Account)

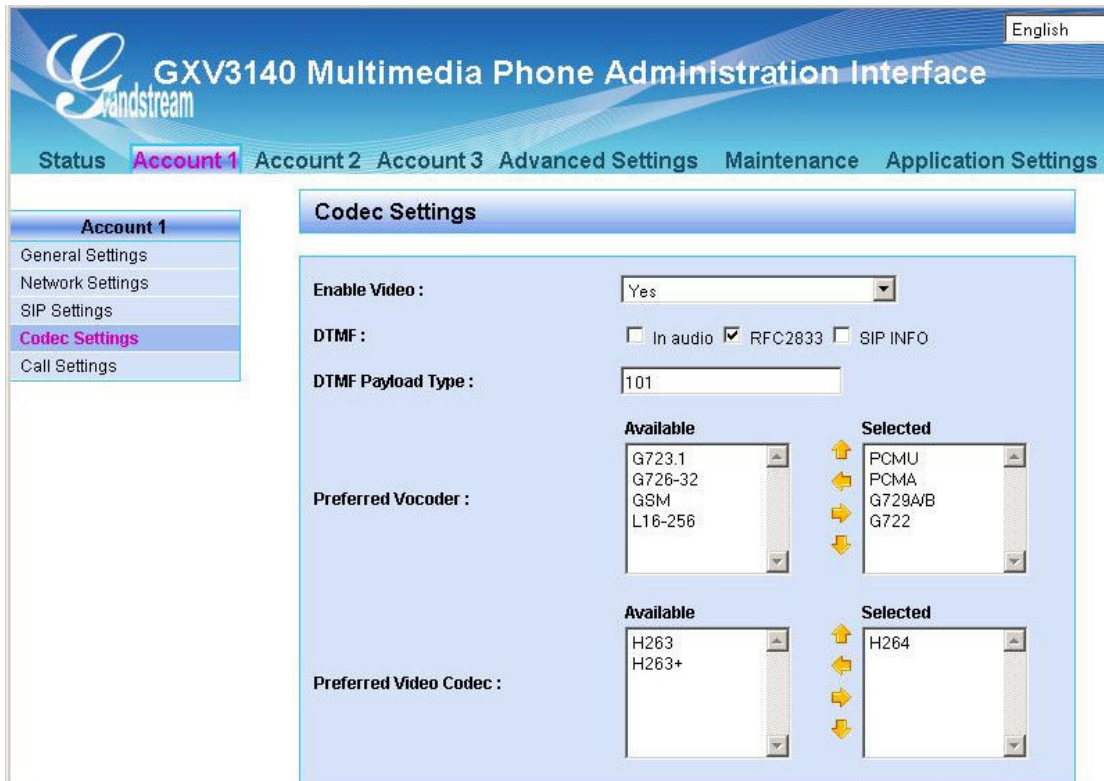
The screenshot displays the 'GXV3140 Multimedia Phone Administration Interface' for 'Account 1'. The left sidebar contains a menu with 'Account 1' selected, and sub-items: 'General Settings', 'Network Settings', 'SIP Settings' (highlighted), 'Codec Settings', and 'Call Settings'. The main area is titled 'SIP Settings' and contains the following configuration options:

- SIP Registration :** ☒ Yes **Set for 1st Account only !**
- Unregister On Reboot :** ☐ Yes
- Register Expiration (m) :** 15
- Wait Time Retry Registration (s) :** 20
- Local SIP Port :** 5060
- SUBSCRIBE for MWI :** ☒ Yes (highlighted with a red box)
- Session Expiration (s) :** 180

Below these settings, there are additional options:

- Min-SE (s) :** 90
- UAC Specify Refresher :** Omit
- UAS Specify Refresher :** UAC
- Force INVITE :** ☐ Yes
- Caller Request Timer :** ☐ Yes
- Callee Request Timer :** ☐ Yes
- Force Timer :** ☐ Yes
- Enable 100rel :** ☐ Yes
- SIP Transport :** UDP
- Symmetric RTP :** ☒ Yes
- Support SIP Instance ID :** ☐ Yes (highlighted with a red box)
- Validate Incoming Messages :** ☐ Yes
- SIP T1 Timeout :** 0.5 sec
- SIP T2 Interval :** 4 sec
- Remove OBP from route :** ☐ Yes

At the bottom right of the settings area are 'Save' and 'Cancel' buttons.



Account 1

- General Settings
- Network Settings
- SIP Settings
- Codec Settings**
- Call Settings

Codec Settings

Enable Video : Yes

DTMF : ☐ In audio ☒ RFC2833 ☐ SIP INFO

DTMF Payload Type : 101

Preferred Vocoder :

Available	Selected
G723.1	PCMU
G726-32	PCMA
GSM	G729A/B
L16-256	G722

Preferred Video Codec :

Available	Selected
H263	H264
H263+	

The dial plan has to be configured as {x+ | *x+} to allow dialling of all strings (default dial plan).

The Refer To Use Target Contact MUST be activated to allow transfer



Account 1

- General Settings
- Network Settings
- SIP Settings
- Codec Settings
- Call Settings**

Call Settings

Dial Plan Prefix :

DialPlan : {x+ | *x+}

Early Dial : ☐ Yes


Refer-To Use Target Contact : ☒ Yes

Auto Answer : No

Send Anonymous : ☐ Yes

Anonymous Call Rejection : ☐ Yes

1.2.2 Hold/Retrieve/Alternate

Hold / retrieve is controlled by a dedicated Key : 

1.2.3 Transfer


Blind - and Attended-Transfer is supported

In Account->Call Settings-> Refer To Use Target Contact MUST be activated to allow Blind transfer

Blind transfer is invoked by pressing  and entering the transfer target.

For invoking Attended-Transfer please refer to the description in the user manual.

Excerpt from manual:

Attended Transfer: Press the "LINE" button () to select an idle line to use for attended transfer; this will place the other party on hold immediately. Dial the number that you wish to transfer to and after confirmation from the party, press the "CALL TRANSFER" button. The phone will display the following message: "Dial Number (Blind) OR Select Line (Attended)". (See figure below). Press the "LINE" button and select the line on hold.

1.2.4 CLIP/CLIR/CNIP - Name and Number presentation

The phone can display names (default) or the call number

Privacy can be activated by feature code and/or Web-interface

Feature Code	Feature
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)

1.2.5 Call Waiting / Call offer

Call waiting is enabled by default in GXV3140 but has to be enabled in OpenScope Office WBM too. As this is a station oriented parameter there is no need to configure it in the phone. Nevertheless two parameters are provided in Web Interface to disable call waiting:



English

GXV3140 Multimedia Phone Administration Interface

Status Account 1 Account 2 Account 3 **Advanced Settings** Maintenance Application Settings

Advanced Settings

- General Settings
- Call Features**
- Video Settings
- Ring Tone

Call Features

Disable Call-Waiting : ☐ Yes

Disable Call-Waiting Tone : ☐ Yes

Disable Direct IP Call : ☒ Yes

Offhook Auto Dial :

Save Cancel

Control of Call Waiting is possible by feature codes as well.

1.2.6 Call Forwarding

The endpoint offers

- CFU **Unconditional Call Forward**
- CFB **Busy Call Forward**
- CFNR **Delayed Call Forward**

Call forwarding is activated/deactivated by feature codes.

Feature Code	Feature
*72	Unconditional Call Forward: Dial *72 + Phone/Ext. Number followed by the # key. Wait for a dial-tone and then hang up (dial-tone means input is successful).
*73	Cancel Unconditional Call Forward: Dial *73 and wait for a dial-tone before hanging up.
*90	Busy Call Forward: Dial *90 + Phone/Ext. Number followed by the # key. Wait for a dial- tone and then hang up.
*91	Cancel Busy Call Forward: dial *91 and wait for a dial-tone before hanging up.
*92	Delayed Call Forward: Dial *92 + Phone/Ext. Number followed by the # key. Wait for a dial-tone and then hang up.
*93	Cancel Delayed Call Forward: Dial *93 and wait for a dial-tone before hanging up.

In addition a configuration via Web-Interface is possible. The timer for CFNR is configurable using the Web-interface only.

1.2.7 Message Waiting

For this feature the “Account Settings”

- Voice Mail UserID: Access number of VM
- Subscribe for MWI

have to be configured.

A waiting message is signaled by the blue LED on top of the phone.


Voicemail access is possible by dedicated key  if the Voice Mail UserID is configured correctly

1.2.8 Distinctive Ringing

Not supported by GXV3140.

The device can configure distinctive ringtones for 3 different caller IDs

1.2.9 Local phone features

GXV3140 offers a local 3 party conference. Active and held call can be connected to a 3 way conference by pressing the  key.

1.2.10 Known limitations and restrictions

Even if the phone comes with a lot of multimedia features and Web application support, the current software has some deficiencies in terms of call and feature handling.

As the phone supports up to 3 lines, features like consultation and conference are implemented by using different lines. It is not possible to invoke such features with only one line.

Thus the user interface for handling such features is rather complex and needs a lot of key presses.

The phone has no easy option to configure the local tones for a specific country.

The phone needs a REBOOT for a lot of configuration changes. As it is not clear which change needs a reboot and which not it is recommended to REBOOT the phone after every configuration.



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