

# OpenScape Business V1

Tutorial

SIP Endpoint Configuration – Grandstream Phones

Version 1.0

# Definitions

## **HowTo**

An OpenScape Business HowTo describes the configuration of an OpenScape Business feature within the OpenScape Office administration. It addresses primarily trained administrators of OpenScape Business.

## **Tutorial**

Within the OpenScape Business tutorials procedures for installation, administration and operation of specific devices, applications or systems, which are connected to OpenScape Business, are described. The tutorial addresses primarily trained administrators of OpenScape Business.

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## Table of History

Date	Version	Changes
2013-06-14	1.0	Initial Creation

# 1 Grandstream phones

## 1.1 Grandstream GXP280



For information see the Grandstream homepage:

[http://www.grandstream.com/products/gxp\\_series/gxp280/gxp280.html](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	
<b>End User Password:</b> <input type="text"/> (purposely not displayed for security protection)				
<b>IP Address:</b> <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:

<b>Time Zone:</b>	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
<b>Daylight Savings Time:</b>	<input type="radio"/> No <input checked="" type="radio"/> Yes
	Optional Rule: 3.2.7.2.0:11.1.7.2.0:60
<b>Time Display Format:</b>	<input type="radio"/> 12 HOUR <input checked="" type="radio"/> 24 HOUR
	<input type="radio"/> Year-Month-Day
<b>Date Display Format:</b>	<input type="radio"/> Month-Day-Year
	<input checked="" type="radio"/> Day-Month-Year
<b>Display Clock instead of Date:</b>	<input type="radio"/> No <input checked="" type="radio"/> Yes

Advanced settings:

Enter the IP-Address of your OpenScape Business as NTP server here:

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

<b>Firmware Upgrade and Provisioning:</b>	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)
<b>Phonebook XML Download:</b>	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
<b>LDAP Directory:</b>	LDAP Script Server Path:	<input type="text"/>
<b>Offhook Auto Dial:</b>	<input type="text"/> (User ID/extension to dial automatically when offhook, max length 35)	
<b>DTMF Payload Type:</b>	<input type="text" value="101"/>	
<b>Onhook Threshold:</b>	<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.

<b>Distinctive Ring Tone:</b>	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"/>
<b>System Ring Tone:</b>	<input type="text" value="f1=440,f2=480,c=200/400;"/>	
<b>Call Progress Tones:</b>	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/offset1[-on2/offset2[-on3/offset3]]]; (Frequencies are in Hz and cadence on and off are in 10ms)		

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

<b>Disable Direct IP Calls:</b>	<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

## Registration and Basic Telephony

### Account settings:

Phone Value	configured in OpenScape Business:
SIP-Server:	IP-Address of OpenScape Business
	configured in OpenScape Business: 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, OpenScape Business uses the name configured in system

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

Adjust the codec settings if needed:

## 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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<b>Send Flash Event:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
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### 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.

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

<b>Disable Transfer:</b>	<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

<b>Display CID instead of Name:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
-------------------------------------	-------------------------------------	---------------------------

Enable CLIR if required, by setting

Send Anonymous                Yes

Anonymous Method            Use Privacy Header

<b>Send Anonymous:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes	(caller ID will be blocked if set to Yes)
<b>Anonymous Method:</b>	<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 Business WBM. As this is a station oriented parameter there is no need to configure it in the phone. Nevertheless two parameters are provided:

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

<b>SUBSCRIBE for MWI:</b>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<b>SUBSCRIBE for Registration Event:</b>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<b>Proxy-Require:</b>	<input type="text"/>	
<b>Voice Mail UserID:</b>	<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](http://www.grandstream.com/products/gxv_series_phone/gxv3140/gxv3140.html)

Used Endpoint:

English

 **GXV3140 Multimedia Phone Administration Interface**

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

Status

Account Status

Network Status

System Info

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 language dropdown set to 'English'. A navigation bar includes 'Status', 'Account 1' (selected), 'Account 2', 'Account 3', 'Advanced Settings', 'Maintenance', and 'Application Settings'. A left sidebar lists 'Account 1' settings: 'General Settings' (selected), 'Network Settings', 'SIP Settings', 'Codec Settings', and 'Call Settings'. The main content area 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 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 same administration interface. The layout is identical to the previous screenshot, but the 'Network Settings' option is selected in the sidebar. The main content area contains the following fields:

- Outbound Proxy :**
- DNS Mode :**
- NAT Traversal :**  (This field is highlighted with a red rectangle)
- Proxy-Require :**

At the bottom 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)

English

## GXV3140 Multimedia Phone Administration Interface

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

**Account 1**

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

### SIP Settings

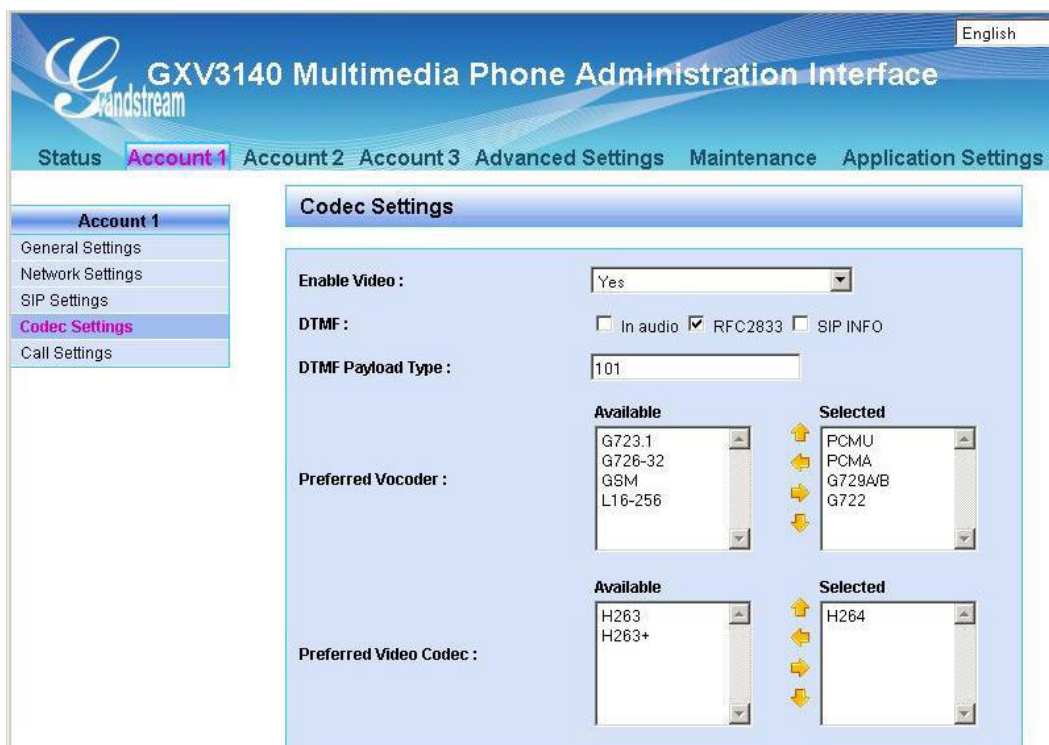
- SIP Registration : ☒ Yes **Set for 1st Account only !**
- Unregister On Reboot : ☐ Yes
- Register Expiration (m) :
- Wait Time Retry Registration (s) :
- Local SIP Port :
- **SUBSCRIBE for MWI : ☒ Yes**
- Session Expiration (s) :

**Account 1**

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

- Min-SE (s) :
- UAC Specify Refresher :
- UAS Specify Refresher :
- Force INVITE : ☐ Yes
- Caller Request Timer : ☐ Yes
- Callee Request Timer : ☐ Yes
- Force Timer : ☐ Yes
- Enable 100rel : ☐ Yes
- SIP Transport :
- Symmetric RTP : ☒ Yes
- **Support SIP Instance ID : ☐ Yes**
- Validate Incoming Messages : ☐ Yes
- SIP T1 Timeout :
- SIP T2 Interval :
- Remove OBP from route : ☐ Yes

Save Cancel



**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	PCM
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 OpenScape Business 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:



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	<b>Unconditional Call Forward:</b> 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	<b>Cancel Unconditional Call Forward:</b> Dial *73 and wait for a dial-tone before hanging up.
*90	<b>Busy Call Forward:</b> Dial *90 + Phone/Ext. Number followed by the # key. Wait for a dial- tone and then hang up.
*91	<b>Cancel Busy Call Forward:</b> dial *91 and wait for a dial-tone before hanging up.
*92	<b>Delayed Call Forward:</b> Dial *92 + Phone/Ext. Number followed by the # key. Wait for a dial-tone and then hang up.
*93	<b>Cancel Delayed Call Forward:</b> 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.



## About Unify

Unify is one of the world's leading communications software and services firms, providing integrated communications solutions for approximately 75 percent of the Fortune Global 500. Our solutions unify multiple networks, devices and applications into one easy-to-use platform that allows teams to engage in rich and meaningful conversations. The result is a transformation of how the enterprise communicates and collaborates that amplifies collective effort, energizes the business, and enhances business performance. Unify has a strong heritage of product reliability, innovation, open standards and security.

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