

ACT P104SLD Phone



This page needs updating. The information contained here relates to sipXecs release 2.6 or 2.8 and is therefore rather old. It is unknown how ACT phones behave with the current sipXecs release 4.4. If you have any information, please help us get this page updated.

How To Configure Advantage Century Telecommunications (ACT) SIP Phones with sipXecs

sipXecs, The open source SIP PBX for Linux can work with all SIP compliant phones, gateways, and applications. As many system features depend on the capabilities of the end points, the selection of phones and gateways should be done by carefully taking into account specific requirements of the intended application and deployment.

Features of ACT P104SLD

Manufacturer Web Site: http://www.act-tel.com.tw/_pg/products/productItem.asp?productKey=39

1. Protocol: SIP(RFC 3261)
2. interface: 1 Ethernet (WAN) & 1 Ethernet (PC)
3. LCD: 2 lines x 16 letters
4. Codec: G.711A-Law/u-Law, G.729AB, G.723.1
5. Voice Quality: VAD, CNG, AEC(G.168),
6. Easy setting by Web browser
7. Multi-user registration
8. Multi-line
9. Auto firmware upgrade (TFTP)
10. 3-way conference
11. Speed Dial
12. Phone Book
13. IP address: Static IP/PPPoE/DHCP
14. NAT traversal: UPnP and STUN
15. Handset/Speaker phone
16. Out-of-band DTMF (RFC 2833)
17. Voice Mail with message waiting indicator
18. Call Forward (Busy, No answer, Unconditional)
19. Call Transfer (Unattended, Blind, Attended)
20. Call Park, Call Pick-up
21. QoS: IEEE802.1p/q
22. Anonymous call block, Caller ID
23. Jitter buffer auto adjustment

sipXecs Configuration

The sipX configuration server currently does not support automatic provisioning of ACT SIP phones. However, they can be manually provisioned using the ACT phone's Web UI.

The configuration was tested with firmware revision 02.08.01

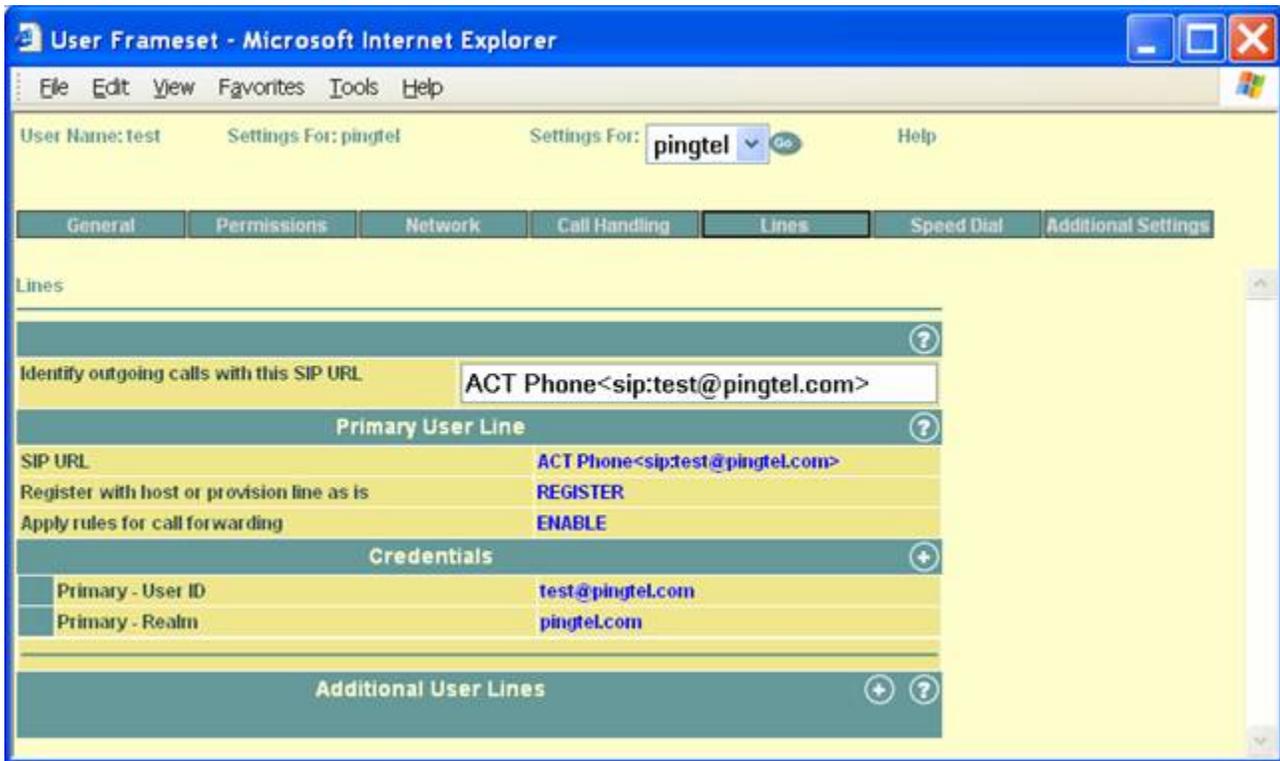
Create a new user in sipXecs

To create a new user in sipX follow these steps:

1. Click 'Add User' button
2. In the sipX Web UI enter 'User ID', 'First', 'Last', 'Extension' and 'PIN'. Also select a user group

Users	Devices	Features	System	Diagnostics
Identification		User: 300		
<ul style="list-style-type: none"> > Phones > Call Forwarding > Schedules > Speed Dial > Group Supervisor > Personal AutoAttendant > Registrations > Permissions > Caller ID 		<p style="text-align: right;"> Show Advanced Settings </p> <p> Existing Groups: <i>administrators, import</i> </p> <p> New Groups: You can create new groups simply by adding the new group name to the Groups form value. </p>		
	User ID	<input type="text" value="300"/>	The User ID can be a numeric extension like "123" or a name like "jsmith". The User ID is displayed by the phone and it is therefore recommended to use the internal extension or the name of the user. If using Direct Inward Dialing (DID), then it is recommended to define the DID number (or its DNIS portion) as an alias.	
	Last name	<input type="text" value="John"/>		
	First name	<input type="text" value="Donovan"/>		
	Active greeting	<input type="text" value="default system greeting"/>		
	E-mail address	<input type="text" value="donovan@example.com"/>	Voicemail prompt callers will hear before leaving a message.	
	Attach voicemail	<input checked="" type="checkbox"/>	Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.	
	Additional E-mail address	<input type="text"/>	If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.	
	Attach voicemail	<input type="checkbox"/>	Used for sending voicemail message notification to the additional e-mail address.	
	PIN	<input type="text" value="*****"/>		
	Confirm PIN	<input type="text" value="*****"/>		
	Groups	<input type="text" value="import"/>	The PIN is a password used to log in to voicemail or to the user portal. Numeric PINs are recommended, since only numbers can be dialed.	
	Aliases	<input type="text" value="donovan"/>	List all groups for this user. If a group does not exist, it will be created. When entering multiple groups, separate them with spaces.	
			Aliases are additional names for the user. Like the user ID, an alias can be either a numeric extension or a name. When entering multiple aliases, separate them with spaces.	
		<input type="button" value="OK"/> <input type="button" value="Apply"/> <input type="button" value="Cancel"/>		

1. Click on the 'Lines' tab to see the format of the line URL, it will usually be username@domain
2. A new device does not have to be created in sipX for the ACT phone, because the phone cannot be configured automatically through sipX configuration server but must be configured using the phone's GUI instead



Configuring the ACT Phone's GUI

Follow these steps to configure the phone:

1. Connect and power up the phone. Default setting is DHCP
2. Press 'Menu', press down arrow a few times to see what the IP address of the phone is, press 'Menu' again to exit
1. Go to http://IP_Address:9999, just hit enter when asked for username/password, both are blank by default
2. Click "SIP Settings" menu in the left frame
3. If DNS SRV is used, enter only 'Registrar Server Domain Name/IP Address'. The domain setting here must be the same as it appears in the 'Lines' tab of the sipX user configuration. If no DNS SRV is used, also enter the hostname of the sipX server in the 'Outbound Proxy Domain Name/IP Address' field. Click 'Submit'.

SIP - Microsoft Internet Explorer

File Edit View Favorites Tools Help

SIP Phone Setting	
SIP Phone Port Number	5060
Registrar Server	
Registrar Server Domain Name/IP Address	pingtel.com
Registrar Server Port Number	5060
Authentication Expire Time	3600 sec. (Default: 3600 sec.)
Outbound Proxy Server	
Outbound Proxy Domain Name/IP Address	
Outbound Proxy Port Number	5060
Others	
Session Timer	1800 sec.
Media Port	41000
Prack	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
Session Refresher	<input checked="" type="radio"/> None <input type="radio"/> UAC <input type="radio"/> UAS
Session Timer Method	<input checked="" type="radio"/> Invite <input type="radio"/> Update

Submit Reset

1. Click 'SIP Account Settings' in the left frame
2. Enter 'Display Name', 'SIP User Name', 'Authentication User Name' and 'Password' for the first account. These setting must be the same as in the 'Lines' tab of the sipX user configuration page. Click 'Submit'.

SIP ACCOUNT - Microsoft Internet Explorer

File Edit View Favorites Tools Help

SIP Account Setting	
Default Account	Account 1
Account 1 Setting	
Account Active	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
Display Name	ACT Phone
SIP User Name	test
Authentication User Name	test@pingtel.com
Authentication Password	1234
Register Status	UnRegister

1. Click 'Voice Settings' in the left frame
2. Set 'DTMF Method' to 'Out Band', click 'Submit'

Microsoft Internet Explorer window titled "Voice". The page contains the following settings:

Voice Setting	
Codec (Priority 1)	G.711 u-law
Codec (Priority 2)	G.729A
Codec (Priority 3)	G.723.1
Codec (Priority 4)	non-used
RTP Packet Length	G.711 μ -Law: 20ms
	G.711 A-Law: 20ms
	G.729A: 20ms
	G.723.1: 30ms
VAD	<input type="radio"/> On <input checked="" type="radio"/> Off
DTMF Method	<input checked="" type="radio"/> Out Band <input type="radio"/> In Band <input type="radio"/> SIP INFO
QoS	
Voice TOS	5 [0 - 7]
Enable/Disable VLAN might Caused Network Connection Problem	
VLAN	<input checked="" type="radio"/> Disable <input type="radio"/> Enable

Buttons: Submit, Reset

1. Click 'Restart System' in the left frame
2. Click 'Restart' button, then click 'Ok' to confirm restart

Microsoft Internet Explorer window titled "Rst". The page displays a red warning message:

Press [Restart] Button, IP Phone system will reboot!

Button: Restart

After restart the phone should register with the sipX server. You should now be able to use it to call other extensions and numbers on registered with sipX as well as receive incoming calls.