

# **Merit LILIN Application Note**

How to Install LILIN IPC0522, Yealink SIP-T19P\_E2 IP Phone and Yeastar S20 SIP Server

Document Number : A00209 Date : 2020/01/15 Dept: Technical Support, Taipei

**Subject:** This document describes how to install and integrate a LILIN IPC0522 with a Yealink SIP-T19P\_E2 IP Phone.

**Device Supported:** LILIN IPC0522 Firmware Version 3.29.92 svn5170M <u>https://www.dropbox.com/sh/xinra4ashmdgdjy/AABoNhJ6aBAhOFfXXhNcwazxa?dl=0</u> Yealink SIP-T19P\_E2 Firmware Version 53.83.0.35 Hardware Version 53.0.0.221.0.0.0 Yeastar S20 Firmware Version 30.10.0.28 Hardware Version V1.30 0000-0000

## How to install:

## Step #1: Yeastar S20 Setting:

• Make sure the S20 and SIP-T19P\_E2 are in the same network segment



• Use default admin/password to login

# **Accessing Web GUI**

Yeastar S-Series provides web-based configuration interface for administrator and extension users. The administrator can manage the device by logging in the Web interface. Check the factory defaults below: IP address: https://192.168.5.150:8088

User Name: <mark>admin</mark>

- Default Password: password
- Settings > PBX > Extensions
  - 1. Add  $\rightarrow$  Checked SIP
  - 2. Extension  ${\scriptstyle \cdot}$  Caller ID  ${\scriptstyle \cdot}$  Registration Name and Caller ID  ${\rightarrow}$  6002
  - 3. Registration Password  $\rightarrow$  Meritlilin988988



	Settings					
	😨 Settings					
20	∨PBX	Extensions Extension	Group			
Configuration	Extensions	Add Bulk Add Edit	Delete Import Export		Extension Name Type	
Wizard	Trunks				Local and the second se	- 0
	Multisite Interconnect		Edit E	xtension ( 6002 )		
	Call Control	Basic Features	Advanced Call Permis	ssion		
Settings	Call Features	General				
	Voice Prompts	Tune			-	
4	General	Type 💽				_
	Recording	Extension ①:	6002	Caller ID 0:	6002	
CDR and	Emergency Number	Registration Name 1:	6002	Caller ID name 1	6002	
Recordings	> System	Concurrent Registrations	1	Registration Password	Meritlilin988988	
3/	> Event Center	Concurrent registrations .		Registration 1 assirona C.		-
×		User Information				
laintenance		Email ①:		User Password ():	********	
		Prompt Language	System Default	Mobile Number		
	13					
				Save Cancel		

# Step #2: Yealink SIP-T19P\_E2 Setting:

• Make sure the SIP-T19P\_E2 and S20 in the same network segment



• Via the phones user interface. Menu > Status to get IP address



• Open a web browser and use the default admin/admin to login

Login	Enterprise IP phone SIP-T19P			
Username	admin			
Password	•••••			
С	onfirm Cancel			



# Account > Register

- 1. Line Active  $\rightarrow$  Enabled
- 2. Display Name  $\sim$  Register Name and User Name  $\rightarrow$  6002
- 3. Password  $\rightarrow$  Meritlilin988988
- 4. Server Host  $\rightarrow$  192.168.110.140

Yealink				English(English) 🗸
	Status Account Net	work Dsskey Fe	eatures Settings	Directory Security
-	Register Status	Registered		NOTE
Register	Line Active	Enabled		NUTE
Basic	Label			Account Registration Registers account(s) for the IP phone.
Codec	Display Name	6002		Server Redundancy
Advanced	Register Name	6002	_	It is often required in VoIP development to ensure service
	User Name	6002		continuity, for events where the server needs to be taken offline
	Password	•••••	-	for maintenance, or for events when the connection between
	SIP Server 1			the IP phone and the server fails.
	Server Host	192.168.5.150	× Port 5060	A computer networking
	Transport	UDP	~	maintaining Internet protocol
	Server Expires	3600		implement NAT.
	Server Retry Counts	3		You can configure NAT traveral
	SIP Server 2			for this account.
	Server Host		Port 5060	The click have to get many
	Transport	UDP	~	product documents.
	Server Expires	3600		
	Server Retry Counts	3		
	Enable Outbound Proxy Server	Disabled	~	
	Outbound Proxy Server 1		Port 5060	
	Outbound Proxy Server 2		Port 5060	
	Proxy Fallback Interval	3600		
	NAT	Disabled	~	
	Confir	m Cancel	]	

### Step #3: IPC0522 Setting

- Login to the IPC0522 Web GUI, Setup > Advance Mode > SIP
  - 1. VOIP  $\rightarrow$  Enable
  - 2. SIP Domain Server  $\rightarrow$  192.168.110.140

  - Register Username → 6000
    Register Password → Meritlilin988988

System Vid	leo / Audio Network	Event Notification Maintenance				
General General IPv6	Advance >> Network >>	SIP				
HTTP/RTSP Service	1 <sub>VOIP</sub>	Enable   Disable				
HTTPS Service	2SIP Domain Server	192.168.110.140				
Multicast	SIP Server Port	5060				
IP Address Filtering	2 Register Lisemame	3000				
DDNS	4 Desister Osernanie	0000				
UPnP	4 Register Password					
Bonjour Service	SIP URL Register Expiration Date	6000@192.168.110.140				
SUDP / Heartbeat	Register Expiration Date	500				
SINIMP	Local SIP Port	5060				
SIF	Audio RTP-Port	7078				
	Video RTP-Port	9078				
	DTMF Receiving	SIP INFO (RFC-2976)				
	DTMF Code (0~9,#,*)	1234#				
	DTMF Time (sec.)	10 🗸				
	Force to end call (sec.)	60 🗸				
		Submit				
	Extension	Ext. 1 V				
	Remote Username	6002				
	SIP URL	6002@192.168.110.140				
	Call Status	Ready				

- Set Extension, and call the SIP phone
  - Remote Username  $\rightarrow 6002$ 1.
  - 2. Click Submit and check Call Status



3. Click Call to connect to Extension 6002 IP phone



## Step #4: DTMF

- SIP server > edit your Extension > Advanced
  - 1. DTMF Mode  $\rightarrow$  Info

		Edit Ex	tension ( 6000 )		
Basic Features	Advanced	Call Permis	sion		
VoIP Settings					,
			🗹 Qualify 🕕		
Register Remotely			Enable SRTP		
T.38 Support ①			DTMF Mode ①:	Info	*
Transport ①:	UDP	*			
Enable User Agent R	egistration Aut	thorization			
Enable User Agent Reg	istration Authoriza	ation ①			
User Agent:			+		
IP Restriction					
Enable IP Restriction	)				
Domitted ID/Ruheet mark:			1	( <del>+</del> )	
			Save Cancel		

- T19P\_E2 > edit account > Advanced
  - 1. DTMF Type  $\rightarrow$  SIP INFO



					Log Out English(English)	
	Status Account Network	Dsskey	Features	Settings	Directory Security	
Denister	Keep Alive Type	Default	~		NOTE	
Register	Keep Alive Interval(Seconds)	30			DTME	
Basic	RPort	Disabled	~		It is the signal sent from the IP	
Codec	Subscription Period(Seconds)	1800			generated when pressing the IP phone's keypad during a call.	
Advanced	DTMF Type	SIP INFO	~			
	DTMF Info Type	DTMF-Relay	~		Session Timer It allows multiple participants	
	DTMF Payload Type(96~127)	101			(more than three) to join a call.	
	Retransmission	Disabled	~		VQ-RTCPXR	
	Subscribe Register	Disabled	~		The VQ-RTCPXR mechanism, complaint with RFC 6035, sends	
	Subscribe For MWI	Disabled	~		the service quality metric reports contained SIP PUBLISH	
	MWI Subscription Period(Seconds)	3600			messages to the central report collector.	
	Subscribe MWI To Voice Mail	Disabled	~			
	Voice Mail				Click here to get more product documents.	

- IPC0522 > edit SIP
  - DTMF Receiving  $\rightarrow$  check SIP INFO DTMF Code (0~9,#,\*)  $\rightarrow$  1234# 1.
  - 2.

System Vid	leo / Audio Network	Event Notification Maintenance				
General	Advance >> Network >>	SIP				
General IPv6						
HTTP/RTSP Service	VOIP	Enable   Disable				
HTTPS Service	SIP Domain Server	192.168.110.140				
Multicast	SIP Server Port	5060				
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	1 DTMF Receiving	SIP INFO (RFC-2976)				
	2DTMF Code (0~9,#,*)	1234#				
	DTMF Time (sec.)	10 🗸				
	Force to end call (sec.)					
		Submit				
	Extension	Ext. 1 🗸				
	Remote Username	6002				
	SIP URL	6002@192.168.110.140				
	Call Status	Ready				

After completing the above settings, you can press the doorbell button and dial the SIP phone to Yealink SIP-T19\_E2 IP Phone. •

Contact

Contact <u>http://lilin.zendesk.com</u> for technical support. For more information, visit www.meritlilin.com.