CASE STUDY

REMOTE RECEPTION OF MEET CALLS IN A PC

Description

This document describes how to configure a MEET panel for receiving calls in a PC using the SIP divert function through the FERMAX MEET ME cloud

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INTRODUCTION

Sometimes we are required for the possibility of receiving calls from a MEET panel in a PC, instead of a MEET monitor specially in offices and/or commercial spaces.

In this Case Study we are going to explain how to implement this function by means of a SIP VoIP client application installed in a PC connected to the internet. This PC can be located in the same building where the panel is or can be installed in any remote location as well.

There exist several client SIP VoIP applications, many of them are free. For this Case Study we are going to use the LINPHONE client APPLICATION, that can be downloaded from the developer web for free under a GNU GPL licence.

Basically this operation consists in the installation of the LINPHONE client in a PC and register it in the FERMAX MEET ME server in the cloud, so the PC receives the calls through this server, in a similar way than a smartphone with the MEET ME APP installed does.

In addition to the MEET panel, a Ref. 1496 MEET ME LICENCE is required for register the PC in the FERMAX MEET ME server.

Calls from the panel can be received in 8 devices simultaneously (PC's and/or smartphones)

HOW TO INSTALL THE LINPHONE CLIENT

Go to <u>www.linphone.org</u> and download the **Linphone-latest-win32.exe** application.

C 🟠 🔒 linphone.org/products			
linphene	Products Solutions Licens	ing & services Technical corner Abou	
	Linphone An open source SIP phone for voice/video calls and instant messaging (mobile and desktop)	Liblinphone A high-level VoIP library implementing all SIP calling and IM features within a single API	Flexisip A SIP server imple proxy, presence a modules
	Mediastreamer2	Belle-sip	ORTP
	A powerful, lightweight streaming engine for voice/video telephony applications	A modern library implementing SIP transport, transaction and dialogue layers	A C library impler Transport Protoco
	bcg729	VoIP tunnel	
			End-to-end encry instant messagin the Signal Protoc
	Since it follows open star telecommunications industry (SIP, interoperable with most PBXs and SIP own Flexisip server, and can be use operator.	Adards from the Many new feature RTP), Linphone is medium- or long-te servers, including our vou want to know d with any SIP VoIP or if you are intere particular feature.	es are still under de rm basis. Please feel fro more about our develo sted in financing the c
	Discover complementary products in	our complete solution	
	for instant messaging and IP-to-IP ca	alls, such as Linphone	

applications, Liblinphone cross-platform VoIP SDK and

Flexisip serve

I. Access to the download section for downloading the application.



II. Select a folder in the PC to save it

💿 Save As						×
← → ` ↑ ↓ > TI	nis PC > Downloads		ٽ ~	,	h Downloads	
Organise 🔻 New fold	ler					?
Quick access Concloud Concloud Concloud Desktop Dosuments Documents Videos Videos Videos Videos Videos USB Drive (D:) dptos (\\srv_nas) (C praxsw (\\srv_nas) (C praxsw (\\srv_nas) (C) Vistor (D:) Vistor (D:)	Name	No itr	Date modified	Туре		Size
	<					>
Save as type: Appl	ication (* exe)					~
Save as type: Appi	ication (iexe)					~
∧ Hide Folders				Save	Can	cel

III. Once downloaded the Linphone-latest-win32.exe file, click on it for installing LINPHONE in your PC.

LINPHONE SETTINGS

REGISTER LINPHONE IN THE MEET ME SERVER

A Ref. 1496 MEET ME LICENCE is required to allow the simultaneous reception of a call in a maximun of 8 different devices (PC's and/or smartphones).

The PC has to be configured and connected to the internet, and the corresponding router will not have any restriction for streaming or SIP protocols. Disable SIP-ALG option if enabled.

The steps are as follow:



I. Arrange the LINPHONE application and use the assistant:

II. select USE A SIP ACCOUNT:

🔄 Linphone		_		\times
A monitor PC sip:linphone@172.30.122.18	Search contact, start a call or a chat	۹	18- 8-8-	Ξ
🛆 номе				
A contacts				
(9) Previously				
	WELCOME			
	This assistant will help you configure and use your sip account.			
		_		
	CREATE A LINPHONE ACCOUNT USE A LINPHONE ACCOUNT			
	USE A SIP ACCOUNT FETCH REMOTE CONFIGURATION	N		

III. Enter the MEET ME domain name (sip.fermax.com) and the Ref. 1496 MEET ME licence credentials (username and password).

Select UDP TRANSPORT PROTOCOL and design a name for it (optional), to identify the account. Finally, select USE to register this account in the MEET ME server.

Cinphone		-		×
A monitor PC sip:linphone@172.30.122.18	Search contact, start a call or a chat	Q	143 444	Ξ
🖒 номе				
A contacts				
Previously	USE A SIP ACCOUNT			
	Username Display name (optional)			
	0995845 PC VIDEO DOORPHONE			
	SIP Domain			
	sip.fermax.com			
	Password			
	þ•••••			
	Transport			
	UDP	~		
	BACK			

IV. Select the new SIP account and check that it is registered. A green dot will appear, otherwise check the entered data in PREFERENCES -> SIP accounts or delete it and create again.



ENABLE H.264 CODEC

Enabling H.264 CODEC in PC is required for a correct performance of the application.

There are the steps to follow:

I. Select "preferences"



II. Select Video \rightarrow H264 \rightarrow OK

	Video input device	Directshow capture:	HP HD Camera		~		
	Video preset	Default			~		
	Video resolution	720p (1280x720)		~			
							VIDEO PREVIEW
ideo code	G						VIDEO PREVIEW
<mark>ideo code</mark> Name	CS Description		Rate (Hz	Bitrat	e (Kbit/s)	Parameters	 VIDEO PREVIEW
<mark>ideo code</mark> Name VP8	CS Description A VP8 video encr	oder using libvpx librar	Rate (Hz) y. 90000	Bitrat	e (Kbit/s)	Parameters	VIDEO PREVIEW Status

*If it is not possible to enable H.264 codec it is because driver is not included during installation process. You must download from http://ciscobinary.openh264.org/openh264-2.1.0-win32.dll.bz2, unzip and move to folder C:\Program Files (x86)\Linphone\plugins\mediastreamer, and rename as openh264.dll.

OTHER CONFIGURATIONS

Door opening in MEET panels requires a "#" DTMF (Dual Tone Multifrequency) digit. Be aware that LINPHONE has this parameter correctly configured and also check that SIP UDP port 5060 is enabled.

Settings							-		×
SIP accounts	🗐 Audio	□1 Video	💪 Calls and Chat	Network	🔞 User Interface	O Advanced			
Transport									
DTMFs s	ending method	SIP INFO	RFC 2833		Allow IPv6	5 💶			
Download speed	limit in Kbit/sec	0		* Upl	oad speed limit in Kbit/se	0		+	
Enable adapt	ive rate control								
Presence									
	Use RLS URI	AUTO	NEVER						
Network prot	ocol and Por	5	Port	Use	a random port	Enabled port			-
	SIP UDP port	5060	*		\bigcirc				
	SIP TCP port	-1	*						
Audi	o RTP UDP port	7078			\bigcirc				
							ок		

MEET DIGITAL OR 1L PANEL SETTINGS

The MEET panel has to be connected to a LAN and have access to the internet. It is not required the panel be connected to the same LAN than the PC is.

This procedure is valid for MILO DIGITAL, MILO 1W, MARINE and KIN panels. It is not valid for KIN panel in 1W PANEL mode.

NETWORK SETTINGS

Configure the panel to have access to the internet, assigning an IP address to it and the corresponding GATEWAY and a valid DNS.

	X 0 DOOR EN	TRY SYSTEM	
DEVICE			NETWORK SETTINGS
GENERAL			
NETWORK	IP:	192.168.1.111	
ACCESS	MASK:	255.255.255.0	
540/41 05000	GATEWAY:	192.168.1.1	
FACIAL RECOG.	DNS:	8.8.8.8	
IP CAMERA	SOFTWARE IP:	192.168.1.200	
SIP	SW. PIN:	•••••	
SIP TRUNK		SAVE	
SIP CALL			
ADVANCED			
PINCODE			
QR ACCESS			
RESET			
LOG OUT			

REGISTER THE PANEL IN THE MEET ME SERVER

the panel should have a MEET ME licence label attached to it.

Please make the following configurations:

FERM	AX		
MEET VIDE	O DOOR ENT	RY SYSTEM	
DEVICE			SIP SETTINGS
GENERAL			
NETWORK	ENABLE SIP:	SEARCH SIP STATUS	SIP REGISTERED
ACCESS	SIP SERVER:	sip:sip.fermax.com	l.
FACIAL RECOG	DOMAIN:	sip.fermax.com	[
FACIAL RECOG.	OUTBOUND:		
IP CAMERA	STUN IP:	192.168.1.233	L
SIP	STUN PORT:	5060	[
	H.264:	102	[
	SIP USER:	0115840	
SIP CALL	SIP PASS:		
ADVANCED	CONVERSATION:	120s 🗸	
PINCODE	RING TIME:	30s 🗸	
QR ACCESS		SAVE	
RESET			
LOCOUT	- 2		

- ENABLE SIP: selected
- SIP SERVER: sip:sip.fermax.com
- DOMAIN: sip.fermax.com
- SIP USER: User indicated in the MEET ME label attached to the panel.
- PASS SIP: password indicated in the MEET ME label attached to the panel.

Click on SIP STATUS to find out if the panel is correctly registered in the server. An indication REGISTERED shall appear. Otherwise verify that the data has been correctly inserted, and the internet configuration is OK.

MAKE A SIP CALL ENTRY

Follow instructions in the panel's INSTALLER MANUAL to create a SIP CALL entry, to say a reference of the number to dial in the panel with the corresponding MEET ME licence registered in the LINPHONE client application, with the format **sip:licence_number@sip.fermax.com.**



In the case of the panel is MEET 1L, configure it as BLOCK. PANEL 1. In this case, when someone press the call button a call to APARTMENT 10001 is always generated.

Finally, create a SIP CALL entry that refers APARTMENT 10001 with the corresponding licence in the LINPHONE client application, whit the format **sip:licence:number@sip.fermax.com-**

	D DOOR ENT	TRY SYSTEM		
DEVICE				SIP CALL SET
GENERAL				
NETWORK	APARTMENT:	0	[
ACC	NUMBER:	_		
SIP	DELETE:	SAVE		
SIP TRUNK		0,02		
SIP CALL	APARTMENT	NUMBER	AP RTMENT NUMBER	APARTMENT
ADVANCED	10001 si	p:0995845@sip.ferma	x.com	
PINCODE				
RESTORE				
LOGOUT				

OPERATION

For receiving calls, the PC has to be ON and with the LINPHONE client application running. t is possible this application to run in background, while the PC is busy with other applications o programs. When someone call from the MEET panel, a ringtone will sound in the PC and a pop up window will appear in the screen.

There are three options:

- Answer with video
- Answer without video (audio only)
- Finish/reject the call



Once a call is answered (with or without audio) a new pop-up will appear with the image from the panel's camera and a panel with several function to use during the conversation (mute, hide image, speaker off, etc.).



- There is possible to hold the conversation by means of the PAUSE icon. The panel will play a "music on hold".
- There is possible to deactivate the outgoing audio (mute) or the incoming audio (speaker).

- The maximum length is defined in the panel configuratio: 30" if no audio has been stablished, and 120" if the call has been answered.
- There is possible to open the door selecting the KEYPAD and then dialling #.

