# CASE STUDY

## RECEPTION OF MEET CALLS IN A PC WITH MICROSIP

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 MicroSIP client APPLICATION, that can be downloaded from the developer web for free under a GNU GPL licence. <u>https://www.microsip.org/</u>

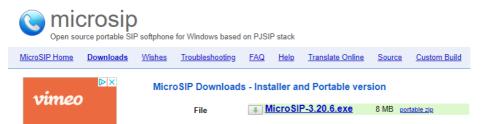
There are two case uses that depend on the communication between panel and PC: first one is when they are located in different LANs and internet is used to communicate each other (SIP mode) and second one when when both are located in the same LAN (P2P mode).

In the first case use MicroSIP must be registered 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 this case, 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 MICROSIP CLIENT

Go to <u>https://www.microsip.org/downloads</u> and download the **MicroSIP-3.20.6.exe** (or currently available) application for Windows OS.



Application will be downloaded automatically in Download folder.

Once downloaded the **MicroSIP-3.20.6.exe** file, click on it for installing MICROSIP in your PC.

## SIP MODE CASE USE (DIFFERENT LAN)

#### **CONFIGURE PANEL**

The steps are as follow: 1) Panel is configured in SIP mode with credential provided in the label sticked on it.

DEVICE		
GENERAL		
NETWORK	ENABLE SIP:	SEARCH SIP STATUS • SIP REGISTERED
ACCESS	SIP SERVER:	sip:sip.fermax.com
	DOMAIN:	sip.fermax.com
FACIAL RECOG.	OUTBOUND:	
IP CAMERA	STUN IP:	
SIP	STUN PORT:	5060
SIP TRUNK	H.264:	102
SIF INONK	SIP USER:	0011825
SIP CALL	SIP PASS:	•••••
ADVANCED	CONVERSATION:	120s 🗸
PINCODE	RING TIME:	30s 🗸
QR ACCESS		SAVE
RESET		

- 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.

After configuration SIP status test is done by clicking on link. SIP REGISTERED in green must be shown.

2) The list of associations between call code and SIP extension is uploaded. For that, an Excel file is created exporting from SIP CALL tab and filling the call code and sip extension separated by ',':

SIP SETTINGS

	A	В	C
1	Apartment,N	lumber,	
2	1,sip:099573	8@sip.ferma	k.com
3	2,sip:1@192.	168.1.220	
4	4,sip:099573	8@sip.ferma	c.com
5			

This Excel file is imported with the IMPORT option:

DEVICE						
GENERAL						
NETWORK	Seleccionar	archivo Ninado IMPORT	EXPORT			
ACCESS	APARTMENT	NUMBER	APARTMENT	NUMBER	APARTMENT	
FACIAL RECOG.	1	sip:0995738@sip.fermax.com	2	sip:1@192.168.1.220	4	sip:0
IP CAMERA						
SIP						
SIP TRUNK						
SIP CALL						
ADVANCED						

If there is a Meet monitor with the same call code (APARTMENT) when calling it will ring in parallel with the SIP extension.

#### **CONFIGURE MICROSIP APP**

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.

1) Create account. Press on down arrow



and select option Add Account

ี้ ও м	icroSIP	–		×		
Teclad	Llamad	das Contactos		~	Activar	
					Editar cuenta	Ctrl+M
					Añadir cuenta	N
1		2 ABC	3 DEF		Ajustes	Ctrl+P
4 <b>4</b> G	н	<b>5</b> JKL	<b>6</b> MNO		Atajos	Ctrl+S
		<b>J</b>	•		Siempre visible	
<b>7</b> P	QRS	<b>8</b> TUV	9wxyz		Ver archivo Log	
*		0	#		lr al sitio Web	Ctrl+W
	_		-		Ayuda	Ver. 3.20.6
	R	+	С		Salir	Ctrl+Q
۲		Llamar	¢	5		
•	-			+		
<u>.</u>	-	1		+		
3		DND AA	CONF R	EC		

Enter MeetMe server and account details. Purchaed MeetMe licence must be used.

Cuenta		$\times$
Nombre de cuenta	МЕЕТМЕ	
Servidor SIP	sip.fermax.com	2
Proxy SIP		2
Usuario*	0995738	2
Dominio*	sip.fermax.com	2
Iniciar sesión		2
Contraseña	******	2
Nombre para mostrar	PC	2
Núm. buzon de voz		2
Prefijo de Marcacion		2
Plan de marcado		2
	Hide Caller ID	2
Comunicacion cifrada	Desactivado $\checkmark$	2
Transporte	UDP ~	2
Dirección pública	Automático ~	2
Refresco de Registro	300 Mantener Conexión 15	
	Publicar presencia	2
	Permitir reescritura IP	2
	ICE	2
	Desactivar temporiz. de sesion	2
x	Guardar Cancelar	

If it is well configured the app shows 'Connected' and SIP extension



#### **AVAILABLE FUNCTIONS**

1) CALL FROM PANEL TO PC

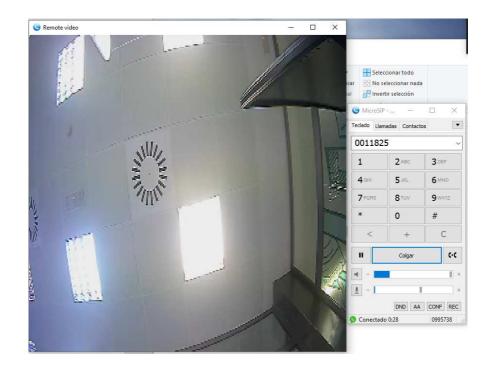
For receiving calls, the PC has to be ON and with the MICROSIP client application running. t is possible this application to run in background, while the PC is busy with other applications o programs. When someone call from the MEET panel, a ringtone will sound in the PC and a pop up window will appear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

There are three options:

- Answer with video
- Answer (audio only)
- Finish/reject the call

Llamada entrante	×
P-1-0-3-DC	OOR ENTRY
0011825@sip DnakeVo	
A: 0995738@si	p.fermax.com
Responder	con video
Responder	Rechazar
cerrar ventana para	ignorar la llamada

Once a call is answered (with or without audio) a new pop-up will appear with the image from the panel's camera and the accessory keypad with several function to use during the conversation (mute, pause, speaker off, etc.). \*Video is shown after 2"



- There is possible to hold the conversation by means of the PAUSE icon.
- There is possible to deactivate the outgoing audio (mute) or the incoming audio (speaker).
- The maximum length is defined in the panel configuration: 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 # or \*.
- There is possible to divert the call to another PC.
- If artifacts are shown on video try to reduce resolution on panel's camera.

#### 2) CALL FROM PC TO PANEL

There are different ways:

i. Select panel from the events list. It is possible to call back to the panel. It is established an audio conversation (no video).

V P-1-0-3-DOOR ENTRY	0011825	8:04:38	2:00
📞 P-1-0-3-DOOR ENTRY	0011825	8:03:54	
V P-1-0-3-DOOR ENTRY	0011825	28/07/2021 14:22:30	0:09

ii. Dialling the SIP extension of the panel.

iii.Creating one name in the contact list:

🌀 Mic	roSIP	-		×
Teclado	Llamadas	Contacto	S	•
Nombre	^		Número	
🔳 Panel			0011825	
<				>
Q				
E Cone	ctado		099573	8

Conversation is limited to 30".

#### 2) SIMULTANEOUS CALL FROM PANEL TO PC AND MEET MONITOR

For this case the monitor must have assigned the apartment number with the same code used in the call.

- Both PC and monitor ring at the same time.
- The first that answers closes the ringing in the other device.
- Conversation and timing are the same of the first case.

## P2P MODE CASE USE (COMMON LAN)

#### **CONFIGURE PANEL**

The steps are as follow:

1) Panel is configured with SIP mode disabled.

DEVICE			
GENERAL			
NETWORK	ENABLE SIP:		
ACCESS	SIP SERVER:	sip:sip.fermax.com	
FACIAL RECOG.	DOMAIN:	sip.fermax.com	
FACIAL RECOG.	OUTBOUND:		
IP CAMERA	STUN IP:		
SIP	STUN PORT:	5060	
SIP TRUNK	H.264:	102	
	SIP USER:	0011825	
SIP CALL	SIP PASS:	••••••	
ADVANCED	CONVERSATION:	120s	~
PINCODE	RING TIME:	30s	~
QR ACCESS		SAVE	

2) Define in SIP CALL option the IPs of receivers assigned to different call numbers. Example 2 in previous Excel.

#### 3) Activate QUICK DIAL in ADVANCED tab with IP of PC:

DEVICE		ADVANCED SETTINGS
GENERAL		
NETWORK	QUICK DIAL:	
ACCESS	URL:	sip:1@192.168.1.220
FACIAL RECOG.	ONU(GPON): MAPPING CALL:	
IP CAMERA	WHITE LIST:	
SIP	DIRECTORY:	Seleccionar archivo Ninado IMPORT EXPORT
SIP TRUNK		
SIP CALL		SAVE
ADVANCED		

#### CONFIGURE MICROSIP

Enable local account Settings page:

Sonido de llamada		x 2	Grabacion de LLamada 📃	C:\Users\carlosf\Desktop	Recordings X
	1			● MP3 ○ WAV	REC
Dispositivo de <mark>l</mark> lamada	Predeterminado	~	Metodo DTMF	Automático 🗸 🗸	
Altavoz	Predeterminado	~	Autorespuesta	Boton de control	~
Microfono	Predeterminado	~	Reenvío de llamadas	No 🗸	0 segu
0 - 0.00	plificar microfono	2	Denegar llamada entrante	Boton de control	~
Códecs disponible	ste de nivel por software es Codecs activados	2	Directorio de usuarios		
Opus 24 kHz G. 722 16 kHz G. 722.1 16 kHz	G.711 A-law G.711 u-law		-		
G.722.1 32 kHz G.723 8 kHz G.729 8 kHz GSM 8 kHz	v Portar codec en llama	adas entrantes	<ol> <li>Manejar botones multii</li> <li>Eventos Sonoros</li> <li>Traer al frente en llam.</li> <li>Pos, aleat, vent, contri LLamada en Espera</li> </ol>	Image: Second	orte para auriculares var archivo log var cuenta local ar informe de error habilitar mensajería
G. 723 8 kHz G. 729 8 kHz GSM 8 kHz	•	adas entrantes	<ul> <li>Eventos Sonoros</li> <li>Traer al frente en llama</li> <li>Pos. aleat. vent. contra</li> </ul>	Image: Second	var archivo log var cuenta local ar informe de error
G.7238 KHz G.7298 KHz GSM 8 KHz WAD 2 GEC Op Camara Codec de video	v	• P • 256	<ol> <li>Eventos Sonoros</li> <li>Traer al frente en llam.</li> <li>Pos. aleat. vent. contra ULamada en Espera</li> </ol>	ada entrante 2 Activ estacion 2 PEnvi	var archivo log var cuenta local ar informe de error habilitar mensajería
G.723 8 KHz G.729 8 KHz GSM 8 KHz VAD 2 ØEC Op Camara Codec de video	v v v v v v v v v v v v v v v v v v v	× P 2	<ol> <li>Eventos Sonoros</li> <li>Traer al frente en llam.</li> <li>Pos. aleat. vent. contra ULamada en Espera</li> </ol>	ada entrante 2 Activ estacion 2 PEnvi	var archivo log var cuenta local ar informe de error habilitar mensajería

A new account is created

Cuenta	×
Nombre de cuenta Local (	call by IP address)
Servidor SIP	2
Proxy SIP	2
Usuario	2
Dominio	2
Iniciar sesión	2
Contraseña	2
Mostrar	contraseña
Nombre para mostrar	2
Núm. buzon de voz	2
Prefijo de Marcacion	2
Plan de marcado	2
Hide	Caller ID 2
Comunicacion cifrada Desact	ivado 🗸 🥇
Transporte UDP	~ 2
Dirección pública Autom	ático 🗸 2
Refresco de Registro 300	Mantener Conexión 15
Publi	icar presencia 2
Perm	nitir reescritura IP 2
ICE ICE	2
Desa	activar temporiz. de sesion 2
G	uardar Cancelar

Main screen shows no connection, just MicroSIP.

SicroSIP 🎯	-	- 1	n x		
Teclado Llamadas	Contactos		•		
			~		
1	2 ABC	<b>3</b> DE	F		
<b>4</b> GHI	5 JKL	<b>6</b> MN	10		
7 PORS	<b>8</b> TUV	<b>9</b> wx	ΥZ		
*	0	#			
R	+	0	2		
<u>a</u>	Llamar		Ģ		
u(· -			+		
<u> </u>			+		
DND AA CONF REC					
MicroSIP					

#### **AVAILABLE FUNCTIONS**

1) CALL FROM PANEL TO PC

Dial code assigned to PC IP address (2 in previous example). Same behaviour as through SIP server.

2) CALL FROM PC TO PANEL

#### PC must be connected only to one network.

Several options:

a. Dialling IP address of panel

S MicroSIF	)		_		×		
Teclado Llan	nadas	Contactos			▼		
192.168.1.214 ~							
1		2 ABC	:	3 DEF			
<b>4</b> GHI	4 GHI 5 JKL		(	<b>6</b> MNO			
7 POR	IS	<b>8</b> TUV	9	<b>9</b> wxyz		<b>9</b> wxyz	
*		0	-	#			
<		+		С			
н		Colgar		(+(	5		
••(· =					+		
<u>•</u> - +							
DND AA CONF REC							
Conectad	o 0:08						

b. Calling back from events calls

S MicroSIP			_		×	
Teclado	Llamadas	Contactos				,
Nombre			Número		Hora	^
<b>\\$</b> 192.1	68.1.214		192.168.	1.214	9:27	
<b>\</b> 10099	03		1009903@	₫192	9:26	
V P-1-0	-3-DOOR EN	TRY	1009903@	<u>∎</u> 192	9:24	
<b>\</b> 1009903		1009903@	9:23			
<b>\</b> 1009903		1009903@	9:23			
<b>\</b> 1009903		1009903@	9:23			
<b>\</b> 1009903		1009903@	₫192	9:22		
P-1-0	-3-DOOR EN	TRY	1009903@	₫192	9:21	
91° n + n	2 0000 EN	TOV	10000024	2102	0.01	

c. Creating one name in agenda with IP of panel

🕒 MicroSIP		_		×	
Teclado	Llamadas	Contactos			•
Nombre	^		Número		Informa
🔳 Panel			192.168.	1.214	

Same behaviour as through SIP server.

## CONCLUSSIONS

MicroSIP provides an easy to use and configure SIP client solution to receive calls from MEET panels. It is light and responds fast.