



# CASE STUDY

## CALL RECEPTION IN SNOM SIP TELEPHONE EXTENSIONS

### [Description](#)

This document describes the way to setup MEET IP intercom system from FERMAX to communicate with IP SNOM Telephone by using SIP protocol.

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## INTRODUCCIÓN

In this Case Study, it is described the way to integrate MEET IP Intercom with SNOM SIP telephones.

Despite it is possible to make a phone call to these telephones through a SIP server, in this integration case we will focus in making a call peer to peer directly from the intercom to the telephone, without the need of a SIP server. In case the telephone is registered in a SIP server, it is advised to review the relative case study of the SIP SERVER in MEET WORKS WITH repository [meet.fermax.com](https://meet.fermax.com)

This integration case is focused in the SNOM D785 Telephone. With other SNOM telephones, the process is similar, specially what is related to MEET IP intercom. For more info related to the SIP phone, it is possible to know more at SNOM SERVICE HUB <https://service.snom.com> , looking for the FAQ “[How To Make Peer-to-Peer IP Calls without a Registrar](#)”

In this case, the configuration to be done will be on a 1 pushbutton MILO MEET intercom panel. Other keypad MEET intercom panels are also compatible with SIP Call.

VERSIONS USED :

- snomD785-SIP 10.1.57.14
- MILO MEET 1 pushbutton panel version 3.0

Direct call is often used in those cases in which due to proximity reasons, the intercom panel and the phone are in the same LAN network and the SIP server is in the cloud. It is not necessary to have internet access to do the call from the intercom to the phone. In addition, the call is faster and reliable since even in the case the internet access is not available, the call will be produced since intercom and telephone are in the same LAN.

As an additional remark, despite we will focus in making a call to a single SIP phone, the intercom outdoor **panel can call simultaneously several destinations or telephones**, following a similar method.

## PRELIMINARY CONSIDERATIONS

Before proceeding to the system configuration, it is required to take into account the following points:

### **IP intercom panel and telephone addressing:**

Both, telephone and intercom, need a fixed IP, in the same local IP network where they are installed. Obviously, these IP addresses must not be used in other IP devices. Can be requested to the network admin. For this case, we will use will the SNOM ip address 192.168.1.100 and the MEET MILO intercom 192.168.1.51 ip address.

## OUTDOOR PANEL INTERCOM SETUP

First of all, all devices should be setup in the same ip network range, as per the selected ip addresses.

We will access to the intercom web server. In the MILO MEET 1 pushbutton panel the default ip address is 10.1.1.2 (in MEET Digital panels with keypad like KIN, MILO, MARINE the default address is 10.1.0.1). We must put the computer network interface in the same range 10.1.1.x

The recommended web browser to use is **Google Chrome**. We will introduce the default IP 10.1.1.2 and the user / password to access. By default **user: admin pass: 123456** .



DEVICE	NETWORK SETTINGS
GENERAL	
NETWORK	
ACC	
SIP	
SIP TRUNK	
SIP CALL	
ADVANCED	
PINCODE	
RESTORE	

  

IP:	192.168.1.51
MASK:	255.255.255.0
GATEWAY:	192.168.1.1
DNS:	8.8.8.8
SOFTWARE IP:	192.168.1.178
SW. PIN:	*****

SAVE

In the NETWORK tab, change the IP to the desired one, in this case 192.168.1.51 and click SAVE. Then we need to change the computer network ip address so that it is in the same network range 192.168.1.x

Now we can access again to the webserver introducing the new ip address 192.168.1.51 in the web browser and we will see again the MILO setup page.

In the GENERAL tab, we will configure that is a 1 single pushbutton panel that calls an apartment (that really do not exist) in the block 1, apartment 101. The pushbutton now is mapped to apartment 101.

It should be selected the TYPE 1W PANEL .

The DEVICE NO will be 1 , to indicate that this is the first panel on the installation. If there were more, we will add consecutive numbers.



DEVICE	GENERAL SETTINGS
GENERAL	
NETWORK	
ACC	
SIP	
SIP TRUNK	
SIP CALL	
ADVANCED	
PINCODE	
RESTORE	

  

TYPE:	1W PANEL
BLOCK:	1
APARTMENT:	101
DEVICE NO.:	1
DEVICE TAG:	VIDEOPORTERO (≤16 CHARACTERS)
LANGUAGE:	ENGLISH
PANEL VOLUME:	5
DOOR OPEN VOICE:	<input checked="" type="checkbox"/>
VIDEO RESOLUTION:	640x480
SIP DIVERT MODE:	PARALLEL CALL
DATE FORMAT:	DD/MM/YYYY
DATE:	03 / 01 / 2018
TIME:	02 : 08 : 55
TIME ZONE:	GMT+01:00

SAVE

Now it is required to associate the pushbutton to an IP address or sip extension to call. We will go to SIP CALL tab and map the intercom pushbutton (assigned previously to apartment 101) to an IP address or SIP extension, that will be the **SNOM** telephone.

In the example, the field APARTMENT will be then 101, and the NUMBER will be the ip call destination, that in this case will be a sip call to ip address 192.168.1.100, assigned to SNOM telephone. The syntax to be used is the following: [sip:192.168.14.1](tel:sip:192.168.14.1)

The screenshot shows the FERMAX MEET VIDEO DOOR ENTRY SYSTEM web interface. On the left is a vertical menu with options: DEVICE, GENERAL, NETWORK, ACC, SIP, SIP TRUNK, SIP CALL (highlighted), ADVANCED, PINCODE, and RESTORE. The main area is titled 'SIP CALL SETTINGS'. It contains form fields for 'APARTMENT' (value: 101) and 'NUMBER' (value: sip:192.168.1.100). There is a 'DELETE' checkbox and a 'SAVE' button. Below these fields is a table with 6 columns: APARTMENT, NUMBER, APARTMENT, NUMBER, APARTMENT, NUMBER. The first row has the value '101' in the first APARTMENT column and 'sip:192.168.1.100' in the first NUMBER column.

APARTMENT	NUMBER	APARTMENT	NUMBER	APARTMENT	NUMBER
101	sip:192.168.1.100				

REMARK: These telephones are SIP, so they can be also registered as an extension of a SIP server. If that was the case, and assuming that the server ip would be 192.168.1.199 and the phone extension registered 1122, the call destination to be used would be then [sip:1122@192.168.1.199](tel:sip:1122@192.168.1.199). Where this syntax corresponds to [sip:sip\\_extension@server\\_ip\\_address](tel:sip:sip_extension@server_ip_address)

## SNOM SIP TELEPHONE CONFIGURATION

The SNOM SIP telephone comes with its default ip.

We need to set the ip address in the desired address (in this case 192.168.1.100). This telephone also has a webserver embedded, so we will connect to it using the same web browser Google Chrome. To access we will use the credentials **user: admin** **pass: 0000**

[←](#)
[→](#)
[↺](#)
No es seguro | 192.168.1.100/info.htm

# System Information

Logout

**Operation**  
Home  
Directory

**Setup**  
Preferences  
Speed Dial  
Function Keys  
Identity 1  
Identity 2  
Identity 3  
Identity 4  
Identity 5  
Identity 6  
Identity 7  
Identity 8  
Identity 9  
Identity 10  
Identity 11  
Identity 12  
Action URL Settings  
Advanced  
Certificates  
Software Update

**Status**  
System Information  
Log  
SIP Trace  
DNS Cache  
Subscriptions  
PCAP Trace  
Memory  
Settings  
Manual

## System Information

Phone Type	snomD785-SIP
MAC Address	0004139AEC53
IP Address	192.168.1.100
IPv6 Address	
Firmware Version	snomD785-SIP 10.1.57.14
Firmware URL	
Production Information	Mac:0004139AEC53;Hardware: D785;Date:07/21;Copyright(C) snom technology GmbH
Uptime	18 days, 0 hours, 47 minutes
LCS	18 days, 0 hours, 45 minutes (0)
Memfree	74040 K
CPU	5.21 5.15 5.10 1/68 645
Bootloader Version	2010.12-00001-gd311851f1

## SIP Identity Status

Identity 1 Status	192.168.1.51@192.168.1.51: Network Failure
Identity 2 Status	
Identity 3 Status	
Identity 4 Status	
Identity 5 Status	
Identity 6 Status	
Identity 7 Status	
Identity 8 Status	
Identity 9 Status	
Identity 10 Status	
Identity 11 Status	
Identity 12 Status	

## Net Port

Connection Type	100 Mbit Full Duplex
Status	connected

## PC Port

Connection Type	
Status	not connected

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## Advanced Settings

Logout

Operation

Home

Directory

Setup

Preferences

Speed Dial

Function Keys

Identity 1

Identity 2

Identity 3

Identity 4

Identity 5

Identity 6

Identity 7

Identity 8

Identity 9

Identity 10

Identity 11

Identity 12

Action URL Settings

Advanced

Certificates

Software Update

Network

Behavior

Audio

SIP/RTP

QoS/Security

Update

Network

IPv6

LLDP

LLDP Network Policy Timeout

DHCP

Options on DHCP=on

Options on DHCP=off

IP Address

Netmask

Host Name

IP Gateway

WLAN

AuthMode

DNS

Domain

DNS Server 1

DNS Server 2

Time

NTP Server

NTP Refresh Time (s)

Time Zone

More Controls ?

on off ?

90 ?

on off ?

1 2 3 4 6 12 15 42 43 51 66 67 ?

43 120 125 ?

192.168.1.100 ?

255.255.255.0 ?

? ?

? ?

off ?

? ?

? ?

? ?

? ?

192.53.103.104 pool.ntp.org ?

3600 ?

+1 Spain (Madrid, Palma) ?

As default, SNOM telephones do not receive direct calls from other SIP devices, so that we need to enable this feature. For that we can follow SNOM recommendations at <https://service.snom.com/> looking for “Peer to Peer IP calls”.

## How To Make Peer-to-Peer IP Calls without a Registrar

1. Login to the **Web UseWeb User Interface** Interface of the phone.
2. Go to **Advanced** → **SIP/RTP** → **SIP** → **Network Identity Port**: 5060
3. Go to **Advanced** → **SIP/RTP** → **SIP** → **Listen TCP Port**: On
4. Click **apply**
5. Go to **Advanced** → **QoS/Security** → **Security** → **Filter Packets From Registrar**: Off
6. Click **apply**
7. If you previously configured an Identity on the phone you may need to disable the Identity by turning it Off.
8. Go to the **Directory** of the phone, from the Web UI, and in the number field enter the SIP URI for the far end device you are trying to call.

**Example:** sip: 192.x.x.x where the 192.x.x.x is the actual Public or Private IP address of the other phone.

9. Set **Number Type**: SIP, **Outgoing Identity**: Active, and then click add.
10. Click **“Save”** and then **“Reboot”** at the top of the page.
11. Repeat Steps 1 – 9 on the other phone.
12. Place a test call by pressing the **“Directory”** key on the phone, select the IP address of the phone you configured in Step 7, and then press the dial key.
13. The call should complete.

The screens where to setup the SNOM parameters are the following:

The screenshot displays the SNOM Web User Interface. On the left, a sidebar menu is organized into three sections: **Operation** (Home, Directory), **Setup** (Preferences, Speed Dial, Function Keys, Identity 1-12, Action URL Settings, Advanced, Certificates, Software Update), and **Status** (System Information, Log, SIP Trace, DNS Cache, Subscriptions, PCAP Trace, Memory, Settings, Manual). The main interface features a top navigation bar with tabs for Network, Behavior, Audio, SIP/RTP, QoS/Security (selected), and Update. Below this, the **Quality of Service** section includes fields for RTP and SIP Type of Service (both set to 160). The **VLAN** section has fields for VLAN Id and Priority for both the phone and PC port. The **IEEE 802.1X Authentication** section includes fields for User and Password. The **Security** section contains several toggle options: Ignore Security Advises, Use Hidden Tags, Restrict URI Queries, Allow CSTA Control, Empty Client Cert, Filter Packets from Registrar (highlighted in yellow and set to 'off'), Authentication for SIP Reboot, Authentication for SIP Check-Sync, and Administrator Mode. The **HTTP Server** section includes fields for User (admin), Password, and Authentication Scheme (set to Digest).



Logout

Operation

Home

Directory

Setup

Preferences

Speed Dial

Function Keys

Identity 1

Identity 2

Identity 3

Identity 4

Identity 5

Identity 6

Identity 7

Identity 8

Identity 9

Identity 10

Identity 11

Identity 12

Action URL Settings

Advanced

Certificates

Software Update

Status

System Information

Log

SIP Trace

DNS Cache

Subscriptions

PCAP Trace

Memory

Settings

Manual

snom

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Network

Behavior

Audio

SIP/RTP

QoS/Security

Update

SIP

Network Identity (Port)

5060

?

SIP T1 (ms)

500

?

Timer Support (RFC4028)

☒ on
☐ off

?

SIP Session Timer (s)

3600

?

SIP Dirty Host TTL (s)

?

SIP Max Forwards

70

?

ENUM Suffix

e164.arpa

?

Retry Interval after Failed Registration (s)

300

?

Use user=phone

☒ on
☐ off

?

Require PRACK

☒ on
☐ off

?

Send PRACK

☒ on
☐ off

?

Offer GRUU

☒ on
☐ off

?

Offer MPO

☐ on
☒ off

?

Use Outbound

☐ on
☒ off

?

Use SIP Compact Headers

☐ on
☒ off

?

Listen on SIP TCP Port

☒ on
☐ off

?

Register HTTP Contact

☐ on
☒ off

?

Disable Blind Transfer (REFER)

☐ on
☒ off

?

Disable Deflection (Code 302)

☐ on
☒ off

?

Show History-Info

☒ on
☐ off

?

Show Diversion

☒ on
☐ off

?

Use NAPTR on SIP URIs

☐ on
☒ off

?

RTCP-XR Report Format

?

Release Transferred Party On

180

?

Retrieve Transferred Party On

400

?

Allow SIP Settings

☐ on
☒ off

?

SIP Header Warning

☐ on
☒ off

?

SIP Header Warning Codes

?

Minibrowser

XML NOTIFY Support

☒ on
☐ off

?

RTP/RTCP

Dynamic RTP Start Port

49152

?

Dynamic RTP End Port

65534

?

DTMF Payload Type

?

RTCP Support

☒ on
☐ off

?

RTP Keepalive

☒ on
☐ off

?

Multicast

Multicast Support

☐ on
☒ off

?

It is convenient to confirm that the MEET compatible codecs are selected. We can do it on the SIP screen. Main codecs are PCMU and PCMA.



# Configuration Identity 1

Logout

**Operation**

- Home
- Directory

**Setup**

- Preferences
- Speed Dial
- Function Keys
- Identity 1
- Identity 2
- Identity 3
- Identity 4
- Identity 5
- Identity 6
- Identity 7
- Identity 8
- Identity 9
- Identity 10
- Identity 11

Login
Features
SIP
NAT
RTP
Audio

**RTP Identity Settings**

Codec: pcmu,pcma ?

Packet Size: 20 ms ?

Filtered Codec List: pcmu, pcma

Full SDP Answer: ☒ on ☐ off ?

Symmetrical RTP: ☐ on ☒ off ?

RTP Encryption: ☒ on ☐ off ?

Enable Mediasec: ☐ on ☒ off ?

Dynamic G.726 Payload: ☒ on ☐ off ?

G.726 Byte Order: ☒ RFC3551 ☐ AAL2 ?

SRTP Auth-Tag: ☒ AES-32 ☐ AES-80 ?

RTP/SAVP: Off ?

Media Transport Offer: UDP ?

Media Transport Offer Setup: active ?

Apply

With the previous configuration we can make a call from the intercom panel to the telephone.

The door lock release can be done using either “\*” or “#” from the telephone keypad, during the call.

## VIDEO

Currently SNOM telephones do not allow video H.264 reception, which is the video sent by MEET FERMAX intercom panels.

## ADVANCED FUNCTIONS

It is possible to call from the telephone to the panel. For that reason, it is quite useful to setup a direct access on the telephone. In particular, SNOM D785 allows to have direct accesses on the auxiliar display.

Additionally, for more usability, it is possible to set up a function key labeled to open the door, in this way even staff that is not familiar with the telephone will be able to identify the way to open the door.



For that it is recommended that a direct access is added to the directory menu.

# Directory

Logout

**Operation**

Home

Directory

**Setup**

Preferences

Speed Dial

Function Keys

Identity 1

Identity 2

Identity 3

Identity 4

Identity 5

Identity 6

Identity 7

Identity 8

Identity 9

Identity 10

Identity 11

Identity 12

Action URL Settings

Advanced

Certificates

Software Update

**Status**

System Information

Log

SIP Trace

**Directory**

Name	Number	Contact Type	Outgoing Identity	Edit	Delete
MILO Panel	192.168.1.51	None	Active		

CSV File
XML File
Delete

**Add or Edit Entry**

Number

Number Type sip

Contact Type None

Outgoing Identity Active

Group None

Title

Organization

Email

Note

Photo Choose File

Action-Url

Nickname

First Name

Family Name

Favorite ☐

Max. 640x480

Add
Add Sub
Change

To label properly and assign the functionality to the quick access function keys, we will access the “Function Keys” and set it up the desired keys.

To **call the outdoor intercom** panel we will choose

Context : Active      Type: Speed Dial      Number: 192.168.1.51      Label : INTERCOM

Where “Number” is the IP of the intercom and “Label” is the text we want to be shown in the function key display.

To **open the door lock during the call** we will choos

Context : Active      Type: DTMF      Number: \*      Label : Door Open

Where “Number” is the DTMF tone to open the door, that in MEET can be “\*” or “#” to active the relay in the intercom panel. In case there is a secondary relay module connected to the panel, then this additional relay can be opened using “0” number. “Label” is the text we want to be shown in the function display.

Some settings are not yet stored permanently.

Save

View Changes

?

Key Settings

On this page you can specify the settings for programmable keys on your phone. Use Context to specify the identity context for that key e.g. this identity will be used to subscribe for a particular extension. Type will select the actual functionality of a particular key. In the last argument field Number, the actual telephone number, sip url, dtmf sequence, action url or key type can be stored. Please refer to your phone manual for more details.

Context-Sensitive Keys

Type	Number	Label	
Key Event	Directory	elldirectory	F1
Key Event	Headset		F2
Call Forward			F3
Key Event	Help		F4

Navigation Keys

Type		
Previous Identity	Up	⬆
Missed Calls	Right	➡
Next Identity	Down	⬇
Accepted Calls	Left	⬅
Redial	OK	✓
Cancel	Cancel	✕

Dedicated Keys

Type	Number		
Key Event	Voicemail	Voicemail	📞
Key Event	DND	DND	🚫
Key Event	Directory	Directory	📖
Transfer		Transfer	📞➡
Key Event	Hold	Hold	⏸
Key Event	Settings	Settings	⚙
Key Event	Next Page	Labels Forward	➡📄
Key Event	Previous Page	Labels Backward	⬅📄

Line Keys

Page Page 1

Context	Type	Number	Label	XML Label
Active	Line			P1
Active	Line			P2
Active	Key Event	Conference		P3
Active	Key Event	Silent Mode		P4
Active	Speed Dial	192.168.1.51	INTERCOM	P5
Active	DTMF	*	Door Open	P6

Apply