

Grandstream HT881 - POTS Gateway

Quick Guide

Version 1.000 (13. February 2025)

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Grandstream HT881

Overview

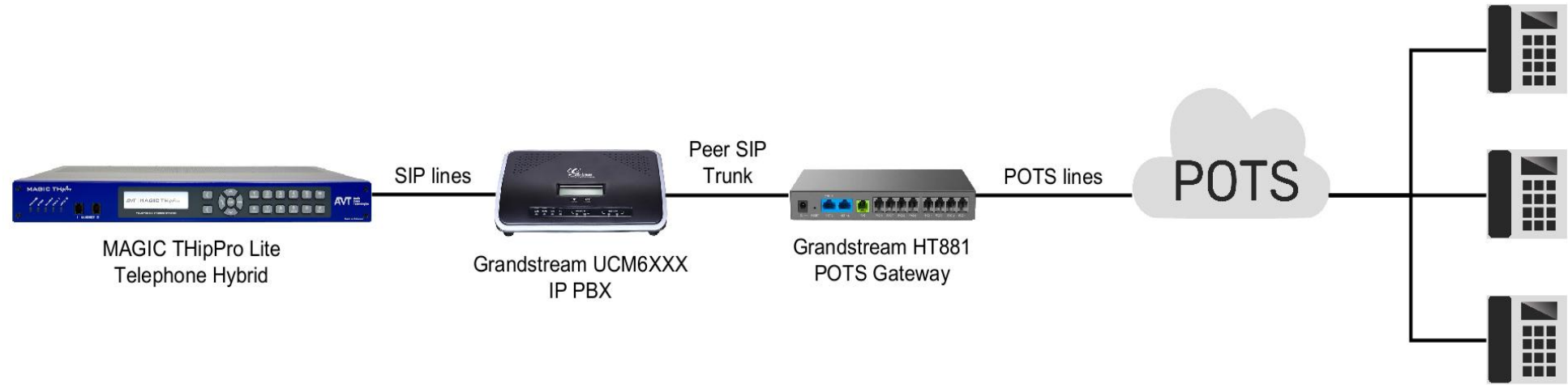
- Hardware
- Application

Hardware – Grandstream HT881



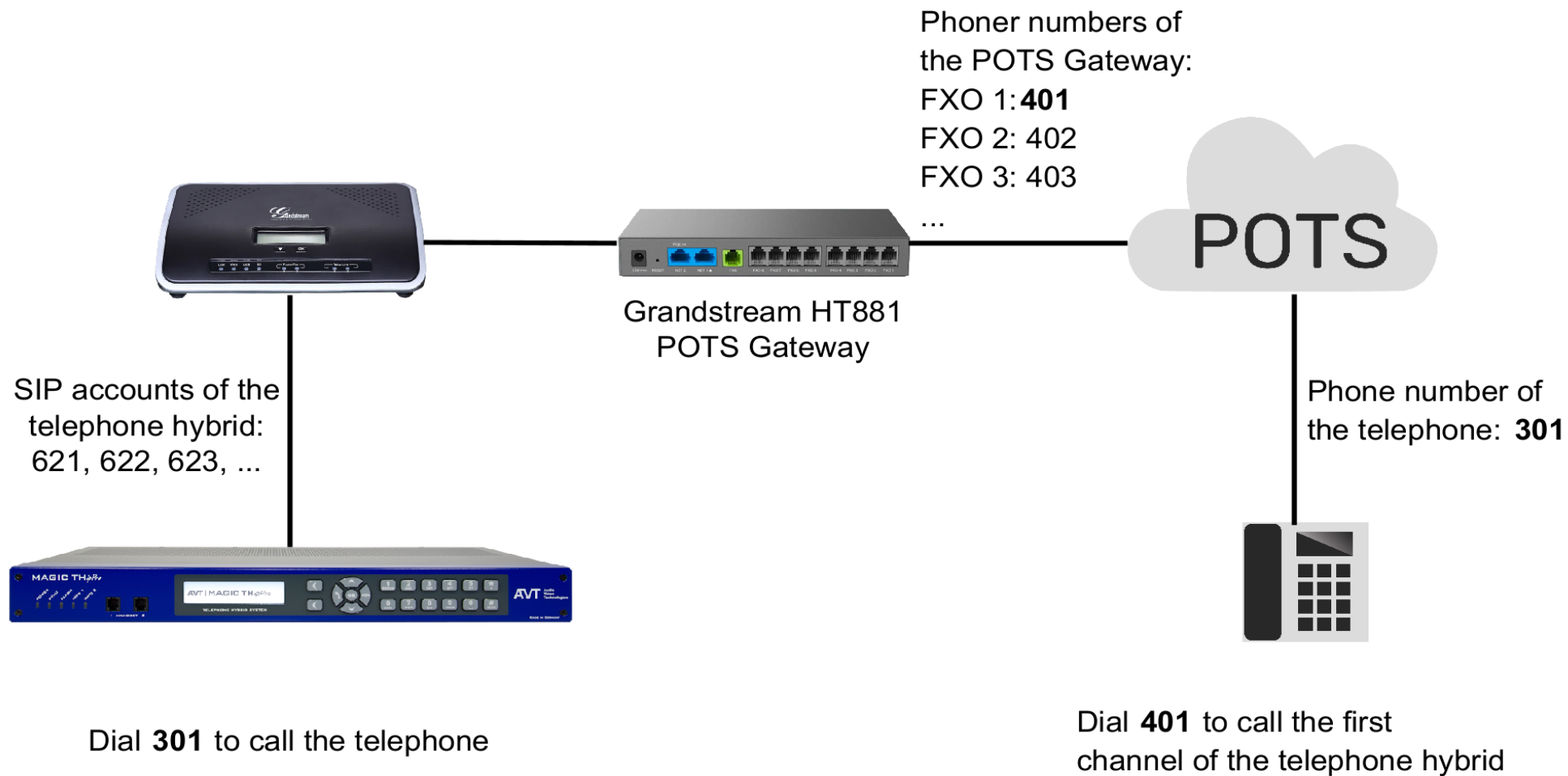
- 8 channel POTS / PSTN gateway
- Does not include a SIP server, a Grandstream UCM6XXX series PBX is required
- Supports only one connection per channel (no call transfer or conferencing)
- A VoIP licence is required on the telephone hybrid (POTS Gateway mode cannot be used)
- A version with 4 channels (HT841) is also available

Application



- The telephone hybrid connects via standard SIP lines to a Grandstream UCM6XXX series PBX
- The IP BPX connects to the Grandstream HT881 POTS gateway via a Peer SIP trunk
- Inbound and outbound routes in the IP PBX interconnect the calls between the telephone hybrid and the POTS gateway
- The Grandstream HT881 POTS Gateway connects via the FXO ports to the POTS network (POTS PBX or provider)

Operation - Example



Grandstream HT881

Configuration of the Grandstream UCM6XXX PBX

- SIP Settings
- Call Termination
- Staging Method

Enter the Web GUI

- Enter that IP address of the UCM6XXX PBX in a web browser to open the Web GUI
- Refer to the User Manual of your specific model for default IP address and login details



SIP Extensions

- Go to **Extension / Trunk** > **Extensions**
- **Add** an extension for each VoIP channel of the Telephone Hybrid

UCM6102

Apply Changes Setup Wizard English admin

Menus

- System Status
- Extension / Trunk
- Extensions
- Extension Groups
- Analog Trunks
- VoIP Trunks
- SLA Station
- Outbound Routes
- Inbound Routes
- Call Features
- PBX Settings
- System Settings
- Maintenance
- CDR
- Value-added Features

Extensions

+ Add Edit Delete Reset Edit All Sip Extensions Import Export E-mail Notification Search

Follow Me Options

<input type="checkbox"/>	Status	Presence Status	Extension	CallerID Name	Message	Terminal Type	IP and Port	Email Status	Options
<input type="checkbox"/>	Idle	Available	621	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.30.15:5060		
<input type="checkbox"/>	Idle	Available	622	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.30.15:5060		
<input type="checkbox"/>	Idle	Available	623	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.30.15:5060		
<input type="checkbox"/>	Idle	Available	624	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.30.15:5060		
<input type="checkbox"/>	Idle	Available	625	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.32.15:5060		
<input type="checkbox"/>	Idle	Available	626	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.32.15:5060		
<input type="checkbox"/>	Idle	Available	627	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.32.15:5060		
<input type="checkbox"/>	Idle	Available	628	MAGIC THipPro Lite	Messages: 0/0/0	SIP(WebRTC)	172.20.32.15:5060		

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New SIP Extension

- Set **Select Extension Type** to **SIP Extension**
- Enter the phone number of the SIP account under **General > Extension**
- Set a password for the SIP account under **SIP/IAX Password**

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Apply Changes Setup Wizard English admin

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- Inbound Routes
- Call Features
- PBX Settings
- System Settings
- Maintenance
- CDR
- Value-added Features

Create New Extension

Basic Settings Media Features Specific Time Follow Me

Select Extension Type: SIP Extension

Select Add Method: Single

General

Extension: 621

Permission: Internal

AuthID:

Voicemail Password: 8247

Send Voicemail to Email: Default

Enable Keep-alive: ☐

Disable This Extension: ☐

Emergency Calls CID:

CallerID Number:

SIP/IAX Password: lm2dp8W

Voicemail: Enable Local Voicemail

Skip Voicemail Password: ☐

Verification:

Keep Voicemail after Emailing: Default

Keep-alive Frequency: 60

Enable SCA: ☐

User Settings

First Name: MAGIC

Last Name: THipPro Lite

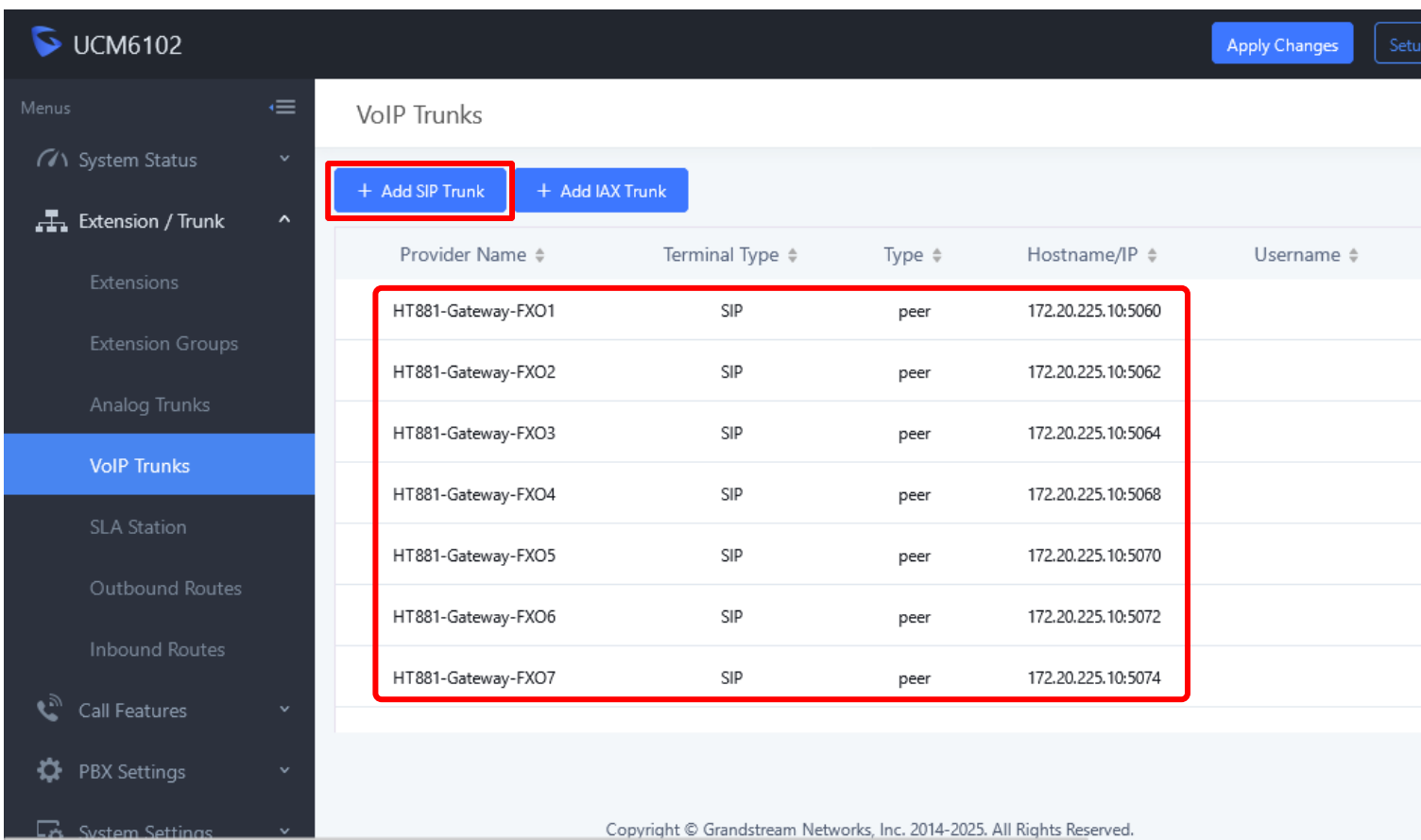
Email Address:

User Password: aHhIL

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Peer SIP Trunks

- Go to **Extension / Trunk** > **VoIP Trunks**
- Add a SIP Trunk for each FXO port of the Grandstream HT881 POTS Gateway



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Apply Changes

Setup

Menus

- System Status
- Extension / Trunk
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- Extension Groups
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- VoIP Trunks**
- SLA Station
- Outbound Routes
- Inbound Routes
- Call Features
- PBX Settings
- System Settings

VoIP Trunks

+ Add SIP Trunk + Add IAX Trunk

Provider Name	Terminal Type	Type	Hostname/IP	Username
HT881-Gateway-FXO1	SIP	peer	172.20.225.10:5060	
HT881-Gateway-FXO2	SIP	peer	172.20.225.10:5062	
HT881-Gateway-FXO3	SIP	peer	172.20.225.10:5064	
HT881-Gateway-FXO4	SIP	peer	172.20.225.10:5068	
HT881-Gateway-FXO5	SIP	peer	172.20.225.10:5070	
HT881-Gateway-FXO6	SIP	peer	172.20.225.10:5072	
HT881-Gateway-FXO7	SIP	peer	172.20.225.10:5074	

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New Peer SIP Trunk

- Set **Type** to **Peer SIP Trunk**
- Set a **Provider Name**
- Set **Host Name** to the **IP address** of the HT881 POTS Gateway and append a specific **port number** for each FXO port using a colon
 - FXO 1: 5060
 - FXO 2: 5062
 - FXO 3: 5064
 - ...
 - FXO 8: 5074

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Apply Changes

Setup

Create New SIP Trunk

Type: Peer SIP Trunk

* Provider Name: HT881-Gateway-FXO1

* Host Name: 172.20.225.10:5060

Keep Original CID: ☐

Keep Trunk CID: ☐

NAT: ☐

Disable This Trunk: ☐

TEL URI: Disabled

Caller ID:

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Outbound Routes

- Go to **Extension / Trunk** > **Outbound Routes**
- Add an Outbound Route for each FXO port of the Grandstream HT881 POTS Gateway

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Apply Changes Setup

Extensions

Extension Groups

Analog Trunks

VoIP Trunks

SLA Station

Outbound Routes

Inbound Routes

Call Features

PBX Settings

System Settings

Maintenance

CDR

Value-added Features

Outbound Routes

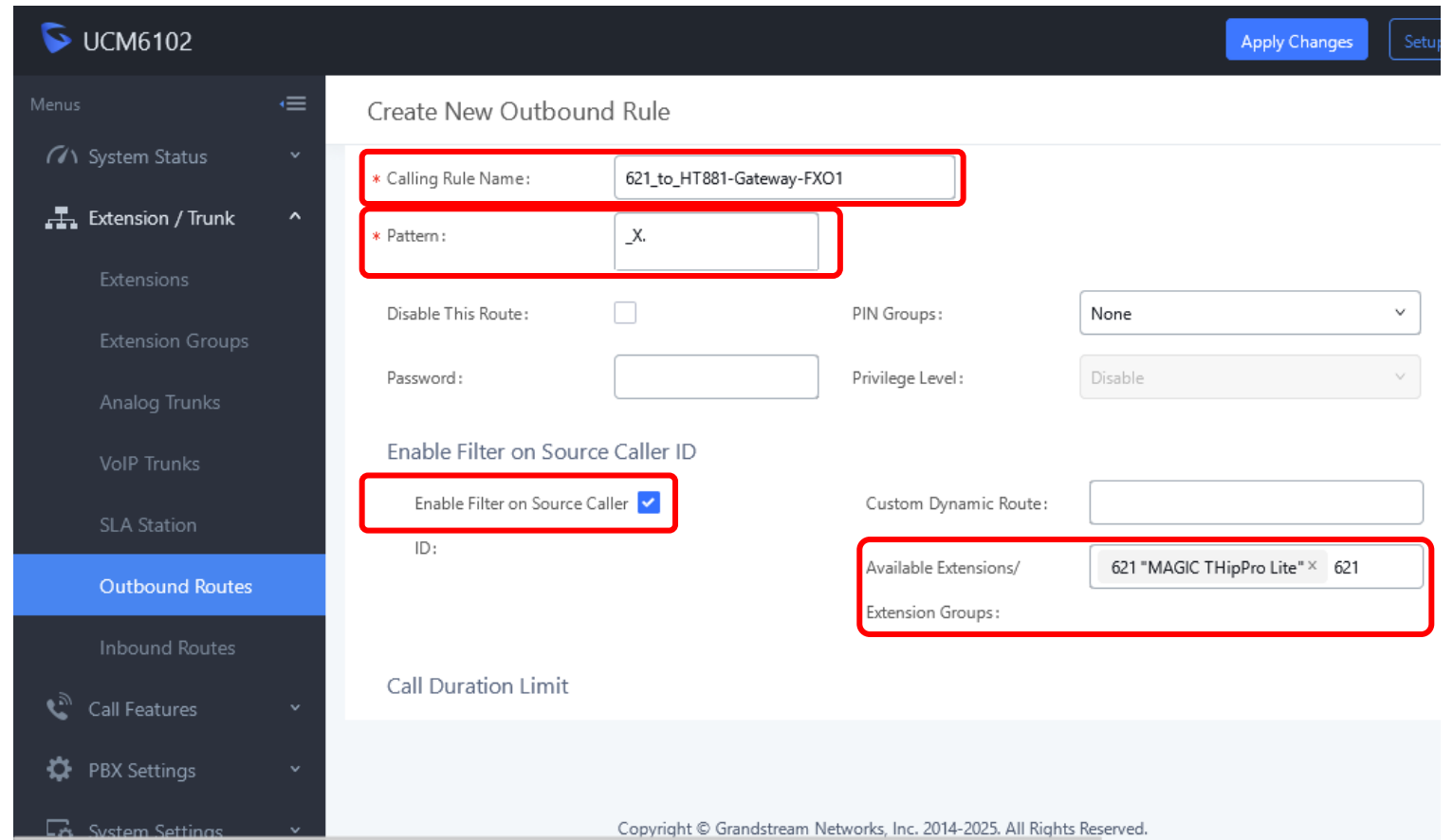
+ Add Outbound Blacklist PIN Groups

Sequence	Outbound Rule Name	Pattern	Privilege Level
1	621_to_HT881-Gateway-FXO1	_X.	Disable
2	622_to_HT881-Gateway-FXO2	_X.	Disable
3	623_to_HT881-Gateway-FXO3	_X.	Disable
4	624_to_HT881-Gateway-FXO4	_X.	Disable
5	625_to_HT881-Gateway-FXO5	_X.	Disable
6	626_to_HT881-Gateway-FXO6	_X.	Disable
7	627_to_HT881-Gateway-FXO7	_X.	Disable
8	628_to_HT881-Gateway-FXO8	_X.	Disable

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New Outbound Route

- Set a **Calling Rule Name**
- Set **Pattern** to **_X.**
- **Enable Filter on Source Caller**
- Under **Available Extensions / Extension Group**, select the SIP Username of the respective line of the Telephone Hybrid



UCM6102

Apply Changes

Setup

Create New Outbound Rule

* Calling Rule Name: 621_to_HT881-Gateway-FX01

* Pattern: _X.

Disable This Route: ☐

Pin Groups: None

Password:

Privilege Level: Disable

Enable Filter on Source Caller ID

Enable Filter on Source Caller ☒

ID:

Custom Dynamic Route:

Available Extensions/Extension Groups: 621 "MAGIC THipPro Lite" x 621

Call Duration Limit

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Inbound Routes

- Go to **Extension / Trunk** > **Inbound Routes**
- Add one Inbound Route for each Peer SIP Trunk

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Apply Changes Setup Wizard English admin




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- Inbound Routes**
- Call Features
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- System Settings

Inbound Routes

+ Add Blacklist Set Global Inbound Mode Import Export

Trunks: SIP Trunks -- HT881-Gateway-FX...

Pattern	CallerID Pattern	Global Inbound Mode	Time Condition	Type	Destination	Options
_621	No Limit	Default Mode	--	--	Default Mode User Extensions -- 621 "MAGIC THipPro Lite 2"	  

Total: 1 < 1 > 10 / page Goto 1

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New Inbound Route

- The Peer SIP Trunk is already selected.
- Enter the phone number of the respective VoIP line of the THipPro as the **Pattern**. Prefix it with an underscore (“_<number>”, in this example: _621).

UCM6102

Apply Changes

Setup

Menus

- System Status
- Extension / Trunk
 - Extensions
 - Extension Groups
 - Analog Trunks
 - VoIP Trunks
 - SLA Station
 - Outbound Routes
 - Inbound Routes**
- Call Features
- PBX Settings
- System Settings

Create New Inbound Rule

* Trunks: SIPTrunks -- HT881-Gateway-FX01

* Pattern: _621

CallerID Pattern:

Prepend Trunk Name:

Alert-info: None

Disable This Route: ☐

Prepend User Defined Name: ☐

Allowed to seamless transfer:

Inbound Multiple Mode: ☐

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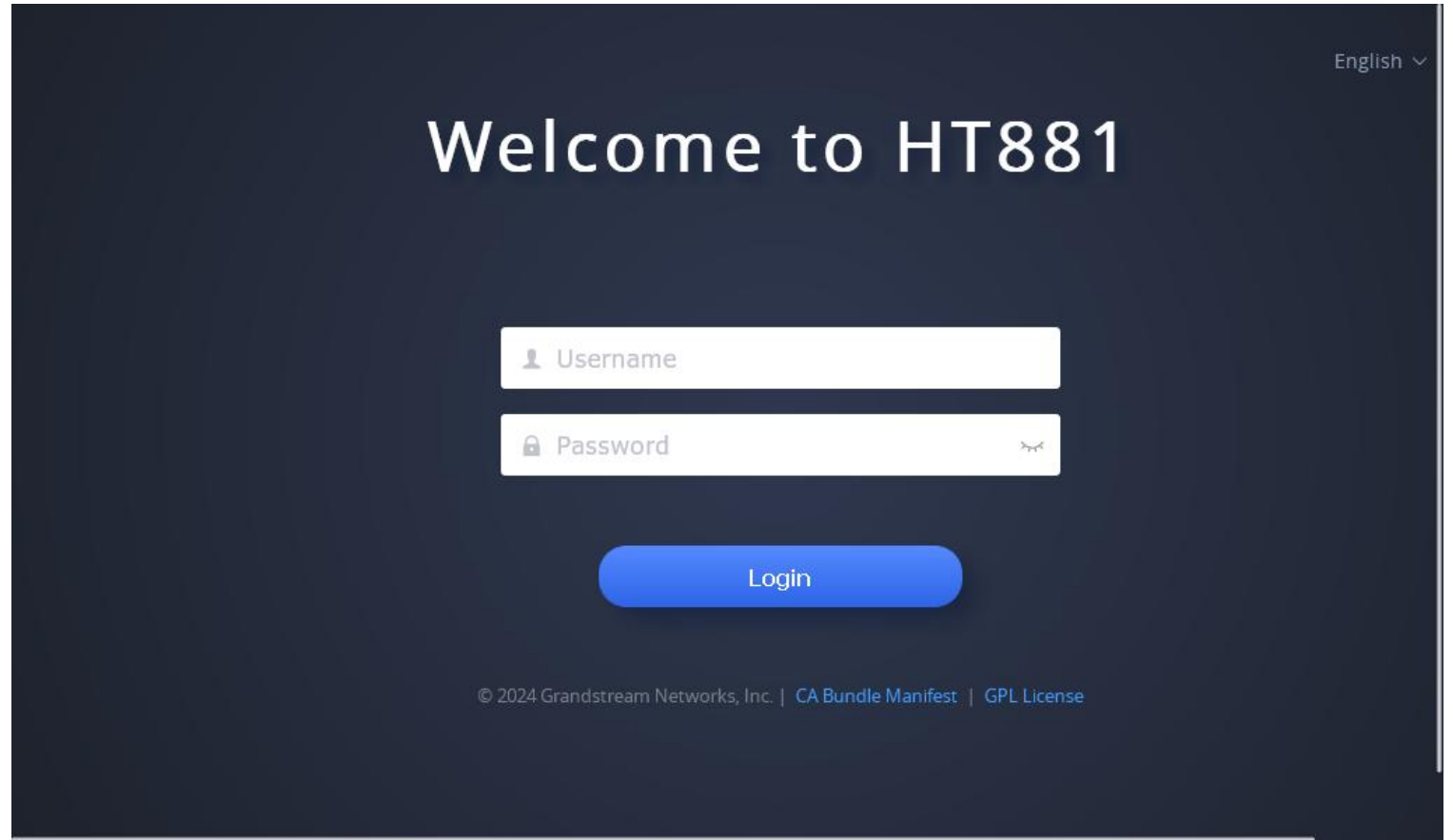
Grandstream HT881

Configuration of the Grandstream HT881

- SIP Settings
- Call Termination
- Staging Method

Enter the Web GUI

- By default, the Ethernet interfaces NET 1 and NET 2 operate in bridge mode
 - NET 1 is the WAN port which will request an IP address via DHCP
 - NET 2 is the LAN port with DHCP pass through from WAN port
- Connect NET 1 of the HT881 to the local network and lookup the IP address it received in the router / DHCP server
- Enter that IP address in a web browser to open the Web GUI
- Default login:
 - Username: admin
 - Password: written on the sticker at the bottom of the device



The screenshot shows the web interface of the HT881 device. At the top right, there is a language selector set to "English". The main heading is "Welcome to HT881". Below this, there are two input fields: "Username" with a user icon and "Password" with a lock icon and a toggle for visibility. A blue "Login" button is positioned below the password field. At the bottom, the footer contains the copyright notice "© 2024 Grandstream Networks, Inc." and links to "CA Bundle Manifest" and "GPL License".

FXS Profile

- Go to **Port Settings > FXS Profile > SIP Settings**
- Change **Local SIP Port** to **6062**

The screenshot shows the HT881 web interface. On the left is a dark sidebar with a menu. The 'Port Settings' section is expanded, and 'FXS PROFILE' is selected. The main content area displays various SIP settings. The 'Local SIP Port' is highlighted with a red rectangular box and is set to the value '6062'. Other settings include 'Delay Time of Port Voltage Off Timer Since Boot' (0), 'Enable SIP OPTIONS/NOTIFY Keep Alive' (No), 'SIP OPTIONS/NOTIFY Keep Alive Interval' (30), 'SIP OPTIONS/NOTIFY Keep Alive Max Lost' (3), 'Local RTP Port' (5004), 'Use Random SIP Port' (No), 'Use Random RTP Port' (No), 'RTP/RTCP Keep Alive On Hold' (No), and 'Hold Target Before Refer' (Yes).

Setting	Value
Delay Time of Port Voltage Off Timer Since Boot	0
Enable SIP OPTIONS/NOTIFY Keep Alive	No
SIP OPTIONS/NOTIFY Keep Alive Interval	30
SIP OPTIONS/NOTIFY Keep Alive Max Lost	3
Local SIP Port	6062
Local RTP Port	5004
Use Random SIP Port	No
Use Random RTP Port	No
RTP/RTCP Keep Alive On Hold	No
Hold Target Before Refer	Yes

FXO Profile 1 – General Settings

- Go to **Port Settings > FXO Profile 1 > General Settings**
- Enter the IP address of the Grandstream UCM 6XXX PBX as **Primary SIP Server**

The screenshot displays the HT881 web interface. On the left is a dark sidebar menu with options: Status, System Settings, Network Settings, Maintenance, Port Settings, FXS PROFILE, FXO PROFILE 1 (highlighted), FXO PROFILE 2, FXS PORT, and FXO PORTS. The main content area is titled 'FXO PROFILE 1' and has tabs for General Settings, SIP Settings, Codec Settings, Call Settings, FXO Termination, and Channel Settings. Under the 'General Settings' tab, the 'Account Registration' section is visible. It includes a 'Profile Active' toggle set to 'Yes', a 'Primary SIP Server' field containing '172.20.20.2' (highlighted with a red box), a 'Failover SIP Server' field, a 'Prefer Primary SIP Server' dropdown set to 'No', an 'Outbound Proxy' field, a 'Backup Outbound Proxy' field, and a 'Prefer Primary Outbound Proxy' toggle set to 'No'.

FXO Profile 1 – SIP Settings

- Go to **Port Settings > FXO Profile 1 > SIP Settings**
- Set **SIP Registration** to **No**

The screenshot displays the HT881 web interface. On the left, a dark sidebar contains a menu with options: Status, System Settings, Network Settings, Maintenance, Port Settings, FXS PROFILE, FXO PROFILE 1 (highlighted), FXO PROFILE 2, FXS PORT, and FXO PORTS. The main content area is titled 'FXO PROFILE 1' and features several tabs: General Settings, SIP Settings (active), Codec Settings, Call Settings, FXO Termination, and Change. Under the 'SIP Basic Settings' section, the 'SIP Registration' option is highlighted with a red rectangular box. It consists of a label 'SIP Registration' followed by a help icon, and two radio buttons: 'No' (which is selected) and 'Yes'. Below this, other settings are visible: 'SIP Transport' with radio buttons for UDP (selected), TCP, and TLS; 'Unregister On Reboot' with radio buttons for No (selected), All, and Instance; 'Outgoing Call without Registration' with radio buttons for No and Yes (selected); 'Register Expiration' with a text input field containing '60'; 'Reregister before Expiration' with a text input field containing '0'; and 'SIP Registration Failure Retry Wait Time' with a text input field containing '20'.

FXO Profile 1 – FXO Termination

- Go to **Port Settings > FXO Profile 1 > FXO Termination**
- Find out if the local POTS network uses **Current Disconnect** or **Disconnect Tone** to disconnect a call and configure the options accordingly

HT881

English admin

FXO PROFILE 1

< General Settings SIP Settings Codec Settings Call Settings **FXO Termination** Change

Enable Current Disconnect ? ☒ No ☐ Yes

Current Disconnect Threshold (ms) ? 100

Enable PSTN Disconnect Tone Detection ? ☐ No ☒ Yes

Enable Polarity Reversal ? ☒ No ☐ Yes

Polarity On Answer Delay ? 1000

AC Termination Model ? ☐ Country-based ☐ Impedance-based

Country-based ? GERMANY

Impedance-based ? 600R -- 600 ohms

PSTN Disconnect Tone

- If Disconnect Tone is used, the tone sequence must be configured under **System Settings > Ringtone > PSTN Disconnect Tone**
- Tone sequences of many countries can be looked up here:

<https://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>

The screenshot shows the HT881 web interface. On the left is a dark sidebar with a menu: Status, System Settings (expanded), Basic Settings, Time and Language, Ringtone (selected), Security Settings, TR069, RADIUS Settings, E911/HELD, Network Settings, and Maintenance. The main content area is titled 'Ringtone' and contains a 'CPT Settings' section. Under this section, there are several configuration fields: 'System Ring Cadence' (c=2000/4000;), 'Prompt Tone Access Code' (empty), 'PSTN Disconnect Tone' (fxsch1-8:f1=425@-32,c=230/230; - highlighted with a red box), 'Dial Tone' (fxsch1:f1=350@-17,f2=440@-17,c=0/0;fxsch1-8:f1=425@-32,c=230/230;), 'Ringback Tone' (fxsch1:f1=440@-17,f2=480@-17,c=2000/4000;), 'Busy Tone' (fxsch1:f1=480@-21,f2=620@-21,c=500/500;fxsch1-8:f1=425@-32,c=230/230;), 'Reorder Tone' (fxsch1:f1=480@-21,f2=620@-21,c=250/250;fxsch1-8:f1=425@-32,c=230/230;), and 'Confirmation Tone' (fxsch1:f1=350@-11,f2=440@-11,c=100/100-100;).

FXO Profile 1 – FXO Termination

- Go to **Port Settings > FXO Profile 1 > FXO Termination**
- Set **Number of Rings** to **2** so, the call will be passed through after the second ringing when the phone number was received.
- Set **PSTN Ring Thru FXS** to **No**.

HT881

English admin

Status

System Settings

Network Settings

Maintenance

Port Settings

FXS PROFILE

FXO PROFILE 1

FXO PROFILE 2

FXS PORT

FXO PORTS

AC Termination Model ☐ Country-based ☐ Impedance-based

Country-based GERMANY

Impedance-based 600R -- 600 ohms

Number of Rings 2

PSTN Ring Thru FXS ☒ No ☐ Yes

PSTN Ring Thru Delay (sec) 4

PSTN Ring Timeout (sec) 6

PSTN Idle Wait Timeout between Outgoing Calls 4

PIN for VoIP-to-PSTN Calls

FXO Profile 1 – Channel Dialling

- Go to **Port Settings > FXO Profile 1 > Channel Dialling**
- Set **Wait for Dial-Tone** to **No** so.
- Set **Stage Method (1/2)** to **1**.

The screenshot displays the HT881 web interface. On the left is a dark sidebar menu with the following items: Status, System Settings, Network Settings, Maintenance, Port Settings (expanded), FXS PROFILE, FXO PROFILE 1 (highlighted in blue), FXO PROFILE 2, FXS PORT, and FXO PORTS. The main content area is titled 'FXO PROFILE 1' and contains several tabs: General Settings, SIP Settings, Codec Settings, Call Settings, FXO Termination, and Channel Dialling (which is the active tab). Under the 'Channel Dialling' tab, the following settings are visible: DTMF Digit Length (ms) set to 100, DTMF Dial Pause (ms) set to 100, First Digit Timeout (sec) set to 10, and Inter-Digit Timeout (sec) set to 4. Two settings are highlighted with red rectangles: 'Wait for Dial-Tone' is set to 'No' (selected with a radio button), and 'Stage Method (1/2)' is set to '1'. At the bottom of the settings list, 'Min Delay Before Dial PSTN Number' is set to 500. At the bottom right of the interface are two buttons: 'Save' and 'Save and Apply'.

Setting	Value
DTMF Digit Length (ms)	100
DTMF Dial Pause (ms)	100
First Digit Timeout (sec)	10
Inter-Digit Timeout (sec)	4
Wait for Dial-Tone	No
Stage Method (1/2)	1
Min Delay Before Dial PSTN Number	500

FXO Ports

- Go to **Port Settings > FXO Ports**
- Enter the SIP Usernames of the respective VoIP lines of the telephone hybrid as **CIDs**
- Set the IP address of the UCM6XXX PBX as **SIP Server** for each port
- Set the **SIP Destination Port** to 5060 for each port

HT881

8

FXO PRO

Disabler

Unconditional Call Forward to VOIP

FXO PORTS	CID	@	Sip Server	:	Sip Destination Port
1	621	@	172.20.20.2	:	5060
2	622	@	172.20.20.2	:	5060
3	623	@	172.20.20.2	:	5060
4	624	@	172.20.20.2	:	5060
5	625	@	172.20.20.2	:	5060
6	626	@	172.20.20.2	:	5060
7	627	@	172.20.20.2	:	5060
8	628	@	172.20.20.2	:	5060

Grandstream HT881

Configuration of the Telephone Hybrid

- Line Interface
- VoIP (LAN / SIP)

Line Interface

- In the PC-Software, go to **Menu > Configuration > System > System Settings > Line Interface**
- Set **Line Mode to VoIP (LAN / SIP)**.
 - Note: POTS Gateway Mode cannot be used with the Grandstream HT881.

The screenshot shows the 'Configuration' window of the Grandstream HT881 software. The left sidebar lists the configuration tree, with 'Line Interface' selected under 'System Settings'. The main panel is titled 'Line Interface' and contains the following sections:

- General:** A dropdown menu for 'Line Mode' is set to 'VoIP (LAN/SIP)' and is highlighted with a red box. Below it are checkboxes for 'Drop not answered incoming/outgoing calls after 90 seconds', 'Pseudo dial tones on current audio interface', 'Disable Lock Function', and 'ECT on PRETALK with Auto Drop'.
- Line Configuration:** A table with 8 channels (1-8) and various line types. The 'In-house Lines' row has all checkboxes checked. Other rows include 'E.164 Lines', 'Collaboration Services', 'Anonymous Calling' (with sip addresses), and 'Reject Anonymous Incoming Calls'.
- PBX/Exchange line configuration:** Fields for 'International prefix' (00), 'National prefix' (0), 'Local Country' (+49), and 'Local Area Code' (0911). It also includes a 'VoIP' section with 'Length of extension' (3), 'Outgoing line prefix' (0), 'PBX number', and a checkbox for 'Skip outgoing line prefix on incoming calls'.

At the bottom right, there are buttons for 'OK', 'Abbrechen', and 'Apply Now'.

VoIP (LAN /SIP)

- In the PC-Software, go to **Menu > Configuration > System > System Settings > VoIP (LAN /SIP)**
- Enter the SIP accounts which were prepared in the Grandstream IP PBX

The screenshot shows the 'Configuration' window of the Grandstream IP PBX software. The left sidebar contains a tree view with the following structure:

- Operation Settings
 - Clients / Security
 - Mode & Audio Line
 - HOLD Signal
 - PhonerSet
 - Signal Processing
 - Monitoring Source
 - Auto Answer
 - Line Labels
 - Database
 - Night Service
 - DTMF Event Labels
 - GPIO
 - Ember+ Consumer Extension
 - Ember+ Dial Pad Extension
- System Settings
 - General
 - Line Interface
 - VoIP (LAN/SIP)** (highlighted)
 - Collaboration Services
 - Audio Interface
 - AES67
 - LAN Interface
 - NTP
 - VLAN
 - DHD Audio Matrix
 - Ember+
 - PhonerSet / Remote Light
 - Stream Quality Measurement
 - SNMP
 - Quick Dials
 - Date and Time
 - Login

The main area displays the 'VoIP (LAN/SIP)' configuration. It features a table with 12 columns: Line, LAN, SIP Server, LAN, Backup Server, TCP, STUN, User Name, User Authenti..., Password, Audio ..., Displayed ..., and DTMF Tx. The table contains 8 rows of data for Line 1 through Line 8. Below the table, there are several configuration panels:

- LAN 1 LAN 2 LAN 3 LAN 4** (tabs)
- STUN Server Parameters**
 - STUN Server: [text box]
 - NAT Keep Alive Message Time: 20 sec (5..60)
- Quality of Service (DiffServ)**
 - Voice: 46 (EF) (0..63) DiffServ: 184dec
 - SIP: 26 (AF 31) (0..63) DiffServ: 104dec
 - Default Settings button
- VoIP Parameter**
 - Payload Time: [slider] 20 msec
 - ☐ A-Law/μ-Law Signalling on incoming G.722 calls
 - ☐ Use first codec of SDP audio codec list as default
 - ☐ Use individual local SIP port numbers Start port: 5062 Show SIP Ports
- Registration**
 - Delay between SIP lines: 0 msec (0..4000)
 - Timeout: 60 sec (60..500)

At the bottom right, there are three buttons: OK, Abbrechen, and Apply Now.



Support Hotline
+49 911 5271 110



Support-Portal
avt-nbg.zammad.com



Support E-Mail
support@avt-nbg.de

Support

A modern, multi-story office building with a glass facade and a prominent overhang. The building is partially covered by a large, semi-transparent red geometric overlay on the right side. The sky is blue with scattered white clouds. A tree is visible in the foreground between the two building sections.

AVT

Audio
Video
Technologies

AVT Audio Video Technologies
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