Grandstream HT813 e FreePBX NethVoice Nethesis

Guida definitiva per Grandstream HT 813 HV 1.1B Firmware 1.0.13.3

La configurazione proposta in questa guida risolve una serie di problemi e permette l'utilizzo corretto di un HT813 con FreePBX 14.0.13 nella distribuzione NethVoice Nethesis. Testato e configurato ed in uso su macchine di produzione.

Le principali problematiche riguardano normalmente, la chiusura della chiamata in ingresso quando il chiamante chiude la linea prima della risposta, e la visualizzazione in chiaro del numero di chi chiama.

1 Configurazione Trunk pjsip sul centralino (Fig.1)

\sim nethvoid	ce		
Admin		Edit Trunk	
Applications	In use by 4 routes		
Connectivity	General Dialed Number Manipulation Rules pjsip Settings		
Dashboard	Trunk Name Ø	cessococool	
Reports	Hide CalleriD ®	Yes No	
Settings	CID Options @		
	Maximum Channels @	Allow Any Clo Block Poreign Clos Remove CNAM Force Trunk Clo	
	Asterisk Trunk Dial Options	T Override System	
	Continue if Busy Ø	Yes No	
	Disable Trunk @	Yes No	
	Monitor Trunk Failures Ø		
		Yes No	

Nel Trunk Name e possibile mettere o il numero o un nome a scelta che identifichi la linea

\sim nethvoice)	
Admin		Edit Trunk
Applications	In use by 4 routes	
Connectivity	General Dialed Number Manipulation Rules	pjsip Settings
Dashboard	JSIP Settings General Advanced Codecs	
Reports	Username	Authentication Disabled
Settings	Secret	Authentication Disabled
	Authentication Ø	Outbound Inbound Both None
	Registration Ø	Send Receive None
	Language Code 🛛	Default
	SIP Server @	192.168.178.160
	SIP Server Port @	5062
	Context Ø	from-pstn
	Transport Ø	0.0.0-udp

Non occorre inserire la Username ed la Secret, è necessario invece il SIP Server che coincide con l'indirizzo IP del HT813 (il dispositivo deve essere collegato tramite la porta WAN e non la porta LAN). La porta SIP Server va impostata come in foto.

Finiti questi passaggi si può salvare e chiudere.

Adesso ci spostiamo sul dispositivo HT813

Il dispositivo va collegato nella porta WAN, se in rete è presente un DHCP potrete raggiungerlo con l'indirizzo che gli verrà assegnato. Di default gli accessi sono user:admin e password:admin, in assenza di DHCP l'indirizzo di default è 192.168.0.1.

Fatto accesso, per prima cosa è necessario impostare l'indirizzo che abbiamo definito in precedenza sul centralino come IP statico del dispositivo.

Tab Basic Setting (Fig. 3)			
Internet Protocol:	● IPv4 Only ○ IPv6 On	ly O Both, prefer IPv4	○ Both, prefer IPv6
IPv4 Address:	O dynamically assigned via I	DHCP	
	DHCP hostname:		(optional)
	DHCP domain name:		(optional)
	DHCP vendor class ID:	HT8XX	(optional)
	○ use PPPoE		
	PPPoE account ID:		
	PPPoE password:		
	PPPoE Service Name:		
	1st Preferred DNS server:	0.0.0	.0
	2nd Preferred DNS server:	0.0.0	.0
	3rd Preferred DNS server:	0.0.0	•0
	4th Preferred DNS server:	0.0.0	•0
	 statically configured as 		
	IP Address:	192 <mark>.</mark> 168 .178	. 160
	Subnet Mask:	255 .255 .255	•0
	Default Router:	192 .168 .178	•1
	DNS Server 1:	192 .168 .178	•1
	DNS Server 2:	0.0.0	.0
IPv6 Address:	• dynamically assigned via I	DHCP	
	○ statically configured as:		
	Full Static		
	Static IPv6 Address:		
	IPv6 Prefix Length:		
	O Prefix Static		
	IPv6 Prefix(64 bits):		
	DNS Server 1:		
	DNS Server 2:		
	Preferred DNS Server:		
Time Zone:	GMT+01:00 (Roma, Paris, Mad	lrid, Prague, Berlin, Budapest, A	Amsterdam, Barcelona) 🗸

La scelta dell'uso del Gateway e del DNS sono necessari per il tipo di configurazione e di risoluzione del centralino in rete, questa scelta condiziona la configurazione della Tab FXO Port dove potremo scegliere se usare l'indirizzo IP del centralino o il nome del dominio.

Se lo volete potete anche impostare la Time Zone.

Scorre la pagina in basso per impostare invece i parametri di configurazione del centralino.

La User ID deve coincidere con il nome del Trunk definito nel centralino Fig. 1, mente per il centralino nel mio caso ho optato per l'indirizzo IP, essendo i due dispositivi nella stessa LAN (Fig. 4)

DMZ IP:				
	WAN port 0	LAN IP	LAN port 0	Protocol
	UDP Only 🗸	·	,	,
	WAN port 0	LAN IP	LAN port 0	Protocol
	UDP Only 🗸			
	WAN port 0	LAN IP	LAN port 0	Protocol
	UDP Only V			
	WAN port 0		LAN port 0	Protocol
Port Forwarding:	UDP Only ▼	LANID	I AN port 0	Dratagal
			LAN port 0	Protocol
	WAN port 0	LANIP	LAN port 0	Protocol
	UDP Only Y			11000001
	WAN port 0	LAN IP	LAN port 0	Protocol
	UDP Only 🗸		, <u> </u>	J
	WAN port 0	LAN IP	LAN port 0	Protocol
	UDP Only 🗸		·	,
Reset Type:	ISP Data Reset 🗸	Reset		
	*00	(V	Manimum Edician T	D.f14 := !!#00!!)
PSIN Access Code:		(Rey pattern to use PSTN mile	e. Maximum 5 digits. I	Jelault is (00)
PIN for VoIP-to-PSTN Calls:	1.0.10	(Maximum 8 digits to authori	ze calling PSTN numb	ers from VoIP. No
	default)			
PIN for PSTN-to-VoIP Calls:		(Maximum 8 digits to authori	ze calling VOIP termin	nals from PSTN. No
5	default)			
Unconditional Call Forward to		(VoIP calls w	vill be forwarded to the	specified PSTN
PSIN:	number)			
	User ID	Sip Server		Sip Destination Port
Unconditional Call Forward to	0808759245	@ 192.168.178.1	:	5160
VOIP:		, = (
	Update	Apply Cancel Reboot		
	All Diabte Dece	nyed Grandetream Networks, Inc. 2006-2021		

Dopo aver fatto Apply, riavviate il dispositivo.

Spostiamoci adesso nella Tab Advanced settings, dove avremo da configurare un pò di cose:

Ecco i parametri da copiare	
System Ring Cadence:	c=1000/4000;
Dial Tone:	f1=425@-12,f2=425@-12,c=200/200-600/1000;
Ringback Tone:	f1=425@-20,c=1000/4000;
Busy Tone:	f1=425@-20,c=500/500;
Reorder Tone:	f1=425@-12,c=250/250;
Call Progress Tones:	
Confirmation Tone:	f1=350@-11,f2=440@-11,c=100/100-100/100;
Call Waiting Tone:	f1=425@-12,f2=425@-12,f3=425@-12,c=400/100-250/100-150/14000;
Prompt Tone:	f1=350@-13,f2=440@-13,c=0/0;
Call Waiting Tone: Prompt Tone:	f1=425@-12,f2=425@-12,f3=425@-12,c=400/100-250/100-150/14000; f1=350@-13,f2=440@-13,c=0/0;

Enable RADIUS Web Access Control:	🔍 No \mid Yes	
Action upon Radius Auth Server Error:	O Reject Access	Authenticate Locally
RADIUS Auth Server Address:		
RADIUS Auth Server Port:	1812	
RADIUS Shared Secret:		
RADIUS VSA Vendor ID:	42397	
RADIUS VSA Access Level Attribute:		
Enable DDNS:	🖲 No 🛛 Yes	
DDNS Server:	dyndns.org 🗸	
DDNS Username:		
DDNS Password:		
DDNS Hostname:		
DDNS Hash:		
System Ring Cadence:	c=1000/4000;	
2,200,000,000,000	(cadence on and off a	re in (0, 16000) ms)
	Dial Tone:	f1=425@-12,f2=425@-12,c=200/200-600/1000;
	Ringback Tone:	f1=425@-20,c=1000/4000;
	Busy Tone:	f1=425@-20,c=500/500;
	Reorder Tone:	f1=425@-12,c=250/250;
Call Progress Iones:	Confirmation Tone:	f1=350@-11,f2=440@-11,c=100/100-100/100-100/100;
	Call Waiting Tone:	[f1=425@-12,f2=425@-12,f3=425@-12,c=400/100-250/100-150/1
	Prompt Tone:	f1=350@-13,f2=440@-13,c=0/0;
	Syntax: f1=val[, f (Frequencies are in (1	2=val[, c=on1/off1[-on2/off2[-on3/off3]]]]; .0, 4000) Hz and cadence on and off are in (0, 64000) ms)
Prompt Tone Access Code		(Key pattern to get Prompt Tone. Maximum 20 digits. No
Trompt Tone Treess Code.	default.)	
Lock Keypad Update:	● No ○ Yes	(configuration update via keypad is disabled if set to Yes)
Disable Voice Prompt:	No O Yes	(voice prompt is disabled if set to Yes)
Disable Direct IP Call:	🔍 No 🛛 🔍 Yes	(direct IP call is disabled if set to Yes)

Configurati questi parametri, si può passare alla Tab FXO Port

I parametri più importanti per questa configurazione sono l'indirizzo IP del centralino o il nome host, non occorre impostare utente o password di autenticazione, ma vi basterà riportare un nome che possa eventualmente identificare la linea PSTN usata, il reso delle configurazini e spunte vanno copiati direttamente come nelle immagini, nella zezione FXO Termination, andronno riportati i dati del paese, ed alcuni parametri fondamentali per la gestione delle chiamate, ecco alcuni parametri da copiare ed usare come proposti:

PSTN Disconnect Tone: f1=425@-12,f2=425@-12,c=200/200-200/200;

	Grandstream Device Configuration
STATUS	BASIC SETTINGS ADVANCED SETTINGS FXS PORT FXO PORT
Account Active:	O No 🔍 Yes
Primary SIP Server:	192.168.178.1 (e.g., sip.mycompany.com, or IP address)
Failover SIP Server:	(Optional, used when primary server no response)
Prefer Primary SIP Server:	O No • Yes (yes - will register to Primary Server if Failover registration expires)
Outbound Proxy:	192.168.178.1 (e.g., proxy.myprovider.com, or IP address, if any)
Backup Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
Prefer Primary Outbound Proxy:	● No ○ Yes (yes - will reregister via Primary Outbound Proxy if registration expires)
SIP Transport:	• UDP O TCP O TLS (default is UDP)
NAT Traversal:	● No O Keep-Alive O STUN O UPnP
SIP User ID:	080XXXXXXXX (the user part of an SIP address)
Authenticate ID:	(can be identical to or different from SIP User ID)
Authenticate Password:	(purposely not displayed for security protection)
Name:	(optional, e.g., John Doe)
DNS Mode	A Decord SDV NADTD/SDV
DNS SRV use Registered IP:	No Ves
Tel URI:	
SIP Registration:	O No 🔍 Yes
Unregister On Reboot:	• No O Yes
Outgoing Call without	O No 💿 Yes
Registration:	(in minutes, default 1 hours man 45 daws)
Register Expiration:	0 (In minutes, detault 1 flour, max 45 days)
SIP Registration Failure Retry	0 (0-04800. Default 0 second)
Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP Registration Failure Retry	1200 (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon
Wait Time upon 403 Forbidden:	403 response.)
Enable SIP OPTIONS Keep Alive:	• No \bigcirc Yes
SIP OPTIONS Keep Alive Interval:	30 (in seconds. Between 1-04800, default is 50)
Enable SIP OPTIONS Keep Alive:	
Endote shi of from the printer	
SIP OPTIONS Keep Alive Interval:	30 (in seconds. Between 1-64800, default is 30)
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max	30 (in seconds. Between 1-64800, default is 30) 3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3.10, default is 3)
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost:	 100 C res 30 (in seconds. Between 1-64800, default is 30) 3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) 26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26)
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS:	30 (in seconds. Between 1-64800, default is 30) 30 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) 26 26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port:	30 (in seconds. Between 1-64800, default is 30) 3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) 26 26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) 5062 (default 5062)
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port:	30 (in seconds. Between 1-64800, default is 30) 3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) 26 26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) 5062 (default 5062) 5012 (even number between 1024-65535, default 5012)
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port:	 100 C res 30 (in seconds. Between 1-64800, default is 30) 3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) 26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) 5062 (default 5062) 5012 (even number between 1024-65535, default 5012) No O Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port:	 No Ves (in seconds. Between 1-64800, default is 30) (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No O Yes No O Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP:	 100 C res (in seconds. Between 1-64800, default is 30) (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes No Yes No Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header:	 100 C res (in seconds. Between 1-64800, default is 30) (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SID Marching	 No Ves
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming	 No No Ites (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE:	 No Ves
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE:	 No Yes (in seconds. Between 1-64800, default is 30) (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) (46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) 5062 (default 5062) (even number between 1024-65535, default 5012) No Yes Yo Yes Yo Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RIP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate	 No Yes (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate domain: Authenticate server certificate	 No Yes (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate domain: Authenticate server certificate chain:	 No Yes (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RIP Port: Use Random SIP Port: Use Random RIP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate domain: Authenticate server certificate chain:	 No No Yes Yes No Yes Yes<!--</th-->
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate domain: Authenticate server certificates: Trusted CA certificates: Allow Incoming SIP Messages from SIP Proxy Only:	 No Yes Yes No Yes Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate domain: Authenticate server certificates: Trusted CA certificates: Allow Incoming SIP Messages from SIP Proxy Only: Use Privacy Header:	 No O Yes
SIP OPTIONS Keep Alive Interval: SIP OPTIONS Keep Alive Max Lost: Layer 3 QoS: Local SIP Port: Local RTP Port: Use Random SIP Port: Use Random SIP Port: Use Random RTP Port: Enable RTCP: Remove OBP from Route Header: Support SIP Instance ID: Validate Incoming SIP Message: Check SIP User ID for incoming INVITE: Authenticate incoming INVITE: Authenticate server certificate domain: Authenticate server certificates: Trusted CA certificates: Allow Incoming SIP Messages from SIP Proxy Only: Use Privacy Header: Use P-Preferred-Identity Header:	 No Ves (in seconds. Between 1-64800, default is 30) (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3) SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) (default 5062) (default 5062) (even number between 1024-65535, default 5012) No Yes Yes No Yes Yes Yes

Use P-Access-Network-Info Header:	○ No ● Yes
Use P-Emergency-Info Header:	O No • Yes
SIP REGISTER Contact Header Uses:	LAN Address O WAN Address
Allow SIP Factory Reset:	• No O Yes
SIP T1 Timeout:	0.5 sec 🗸
SIP T2 Interval:	4 sec 🗸
SIP Timer D:	0 (0 - 64 seconds. Default 0)
DTMF Payload Type:	101
Preferred DTMF method	Priority 1: RFC2833 V
(in listed order):	Priority 2: SIP INFO V
Disable DTMF Negotiation:	No (pagetista with page) O Vas (use above DTME order without pagetistion)
Generate Continuous REC2833	• No (negotiate with peer) • res (use above D rivir order without negotiation)
Events:	• No • Yes (RFC2833 events are generated until key is released)
Flash Digit Control:	• No O Yes (Overrides the default settings for call control when both channels are in use.)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)
SIP User-Agent:	
SIP User-Agent Postfix:	
Do Not Escape '#' as %23 in SIP URI:	● No ○ Yes
Disable Multiple m line in SDP:	• No O Yes
Ring Timeout:	60 (0-300, default is 60 seconds, 0 means no timeout)
Early Dial:	• No O Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	(this prefix string is added to each dialed number)
Use # as Dial Key:	O No • Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan:	{ x+ \+x+ *x+ *xx*x+ }
SUBSCRIBE for MWI:	No, do not send SUBSCRIBE for Message Waiting Indication
	Ves send periodical SUBSCRIBE for Message Waiting Indication

Anonymous Call Rejection:	• No O Yes
Special Feature:	Standard
Session Expiration:	180 (90-64800. default 180 seconds)
Min-SE:	90 (90-64800. default 90 seconds)
Caller Request Timer:	• No • Yes (Request for timer when making outbound calls)
Callee Request Timer:	• No • Yes (When caller supports timer but did not request one)
Force Timer:	• No • Yes (Use timer even when remote party does not support)
UAC Specify Refresher:	O UAC O UAS • Omit (Recommended)
UAS Specify Refresher:	• UAC O UAS (When UAC did not specify refresher tag)
Force INVITE:	• No O Yes (Always refresh with INVITE instead of UPDATE)
INVITE Ring-No-Answer Timeout	40 (5-300 seconds, Default 40 seconds)
(sec):	
Enable 100rel:	• No O Yes
Add Auth Header On Initial REGISTER	● No ○ Yes
REGISTER.	
Use First Matching Vocoder in	No O Yes
2000K SDP:	
Preferred Vocoder	choice 1: PCMU V
(in listed order).	choice 3: G723 V
	choice 4: G729 🗸
	choice 5: G726-32 ✓
	choice 6: iLBC •
Voice Frames per TV:	
G723 Rate:	6 3 Johns encoding rate 0 5 3 Johns encoding rate
iI BC Frame Size:	20ms 30ms
Disable OPUS Stereo in SDP:	No Ves (removes "/2" from offer)
iLBC Pavload Type:	97 (between 96 and 127, default is 97)
OPUS Payload Type	123 (between 96 and 127, default is 123)
VAD.	• No • Ves
Symmetric RTP	\circ No \circ Yes
Fax Mode	• T 38 O Pass-Through
1 at 10000.	• 1.50 • Tass Inforgn

Fax Mode:	● T.38 ○ Pass-Through
Re-INVITE After Fax Tone Detected:	• Enabled O Disabled
Jitter Buffer Type:	O Fixed • Adaptive
Jitter Buffer Length:	○ Low ● Medium ○ High
SRTP Mode:	Disabled Enabled but not forced Enabled and forced
Crypto Life Time:	O Disabled
Caller ID Scheme:	Bellcore/Telcordia
DTMF Caller ID:	Start Tone Default Stop Tone Default
FSK Caller ID Minimum RX Level (dB):	-40 (-96 - 0dB. Default -40dB)
FSK Caller ID Seizure Bits:	70 (0 - 800 bits. Default 70)
FSK Caller ID Mark Bits:	40 (1 - 800 bits. Default 40)
Caller ID Transport Type:	Relay via SIP From
Send Hook Flash To PSTN:	○ No ● Yes (If Yes, hook flash will be sent to PSTN upon receiving flash event from RFC2833 or SIP INFO)
Hook Flash Duration (ms):	600 (200 - 1500 milliseconds. Default 600)
Gain:	<i>TX</i> OdB default ♥ <i>RX</i> -2dB default ♥
Disable Line Echo Canceller (LEC):	• No O Yes
Disable Network Echo Suppressor:	• No O Yes
Outgoing Call Duration Limit:	0 (0-180 minutes, default is 0 (No Limit))
FXO Termination	
Enable Curr Disconne	ent O No • Yes (Default Yes. If set to yes, enter threshold below)
Current Disconr Threshold (n	ect 400 (50-800 milliseconds. Default 100 milliseconds)
Enable PSTN Disconn Tone Detecti	ect O No O Yes (Default No)
	(If set to yes, the following tone is used as the disconnect signal)
PSTN Disconnect To	ne: [f1=425@-12,f2=425@-12,c=200/200-200/200;
	(Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;)

	(Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;)
	(Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm)
	(Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)
Enable Polarity Reversal:	○ No [●] Yes (Default No. Check with your PSTN carrier before setting to Yes)
AC Termination Model	O Country-based O Impedance-based Auto-Detected
Country-based	ITALY
Impedance-based	600R 600 ohms
Number of Rings:	2 (1-50. Default 4) (Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)
PSTN Ring Thru FXS:	• No O Yes (Default Yes)
0	(If set to ves, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)
PSTN Ring Thru Delay (sec):	4 (1-10 seconds. Default 4 seconds)
PSTN Ring Timeout (sec):	6 (2-10 seconds. Default 6 seconds)
	(Used to detect PSTN hangup when FXO port is not answered)
PSTN Idle Wait Timeout between Outgoing Calls:	4 (0-10 seconds. Default 4 seconds)
Channel Dialing	
DTMF Digit Length (ms):	100 (40-127 milliseconds, Default 100 milliseconds)
DTMF Dial Pause (ms):	100 (40-127 milliseconds, Default 100 milliseconds)
First Digit Timeout (sec):	10 (1-20 seconds. Default 10 seconds)
Inter-Digit Timeout (sec):	4 (1-15 seconds. Default 4 seconds)
Wait for Dial-Tone:	O No • Yes (Default Yes - dial upon dial-tone)
Stage Method (1/2):	1 (Default 2 - 2 stage dialing)
Min Delay Before Dial PSTN Number:	500 (default 500ms, range 50 ~ 65000ms)
	Update Apply Cancel Reboot

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Alla fine della configurazione di questa pagina accorre salvare e riavviare l'apparato.

Per concludere adesso vediamo come configurare anche i parametri della porta FXS dal tab FXS Port

Per questa sezione chiaramente i parametri da utilizzare dipendono dal tipo di configurazione del centralino, porte, protocollo e certificato definiscono il tipo di configurazione. Nell'esempio proposto, il centralino usa un certificato ed è raggiungibile anche dall'esterno. La porta utilizzata verrà modificata automaticamente dall'apparato in funzioni del tipo di porta e protocollo.

Non vi preoccupate se nella Tab Status le due porte risultano come Not Registered, fate prima una prova e se avete configurato tutto come descritto, vedrete che funziona perfettamente.

	Grandstream Device Configuration
STATUS	BASIC SETTINGS ADVANCED SETTINGS FXS PORT FXO PORT
Account Active:	O No • Yes
Primary SIP Server:	host/indirizzo ip centralino (e.g., sip.mycompany.com, or IP address)
Failover SIP Server:	(Optional, used when primary server no response)
Prefer Primary SIP Server:	● No ○ Yes (yes - will register to Primary Server if Failover registration expires)
Outbound Proxy:	host/indirizzo ip centralino (e.g., proxy.myprovider.com, or IP address, if any)
Backup Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
Prefer Primary Outbound Proxy:	• No O Yes (yes - will reregister via Primary Outbound Proxy if registration expires)
Allow DHCP Option 120 (override SIP server):	• No O Yes
SIP Transport:	O UDP O TCP I TLS (default is UDP)
SIP URI Scheme When Using TLS:	○ sip ● sips
Use Actual Ephemeral Port in Contact with TCP/TLS:	• No O Yes
NAT Traversal:	No Keep-Alive O STUN O UPnP
SIP User ID:	91105 (the user part of an SIP address)
Authenticate ID:	91105 (can be identical to or different from SIP User ID)
Authenticate Password:	(purposely not displayed for security protection)
Name:	assistenza (optional, e.g., John Doe)
DNS Mode:	A Record O SRV O NAPTR/SRV
DNS SRV use Registered IP:	• No O Yes
Tel URI:	Disabled V
SIP Registration:	O No Ves
Unregister On Reboot:	• No O Yes
Outgoing Call without Registration:	O No O Yes
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
Reregister before Expiration:	0 (0-64800. Default 0 second)

SLIC Setting:	EUROPEAN CTR21
Caller ID Scheme:	Bellcore/Telcordia
DTMF Caller ID:	Start Tone Default V Stop Tone Default V
Polarity Reversal:	• No O Yes (reverse polarity upon call establishment and termination)
Loop Current Disconnect:	• No O Yes (loop current disconnect upon call termination)
Play busy/reorder tone before Loop Current Disconnect:	• No • Yes (play busy/reorder tone before loop current disconnect upon call fail)
Loop Current Disconnect Duration:	200 (100 - 10000 milliseconds. Default 200 milliseconds)
Enable Pulse Dialing:	No O Yes
Pulse Dialing Standard:	General Standard
Enable Hook Flash:	O No 🔍 Yes
Hook Flash Timing:	In 40-2000 milliseconds range, minimum: 300 maximum: 1100
On Hook Timing:	400 (In 40-2000 milliseconds range, default is 400)
Gain:	<i>TX</i> 0dB default ∽ <i>RX</i> -6dB default ∽
Disable Line Echo Canceller (LEC):	• No O Yes
Disable Network Echo Suppressor:	● No ○ Yes
Outgoing Call Duration Limit:	0 (0-180 minutes, default is 0 (No Limit))
Enable High Ring Power:	• No O Yes
RFC2833 Events Count:	8 (between 2 and 10, default is 8)
RFC2833 End Events Count:	3 (between 2 and 10, default is 3)