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CONFIGURAZIONE TURRIS

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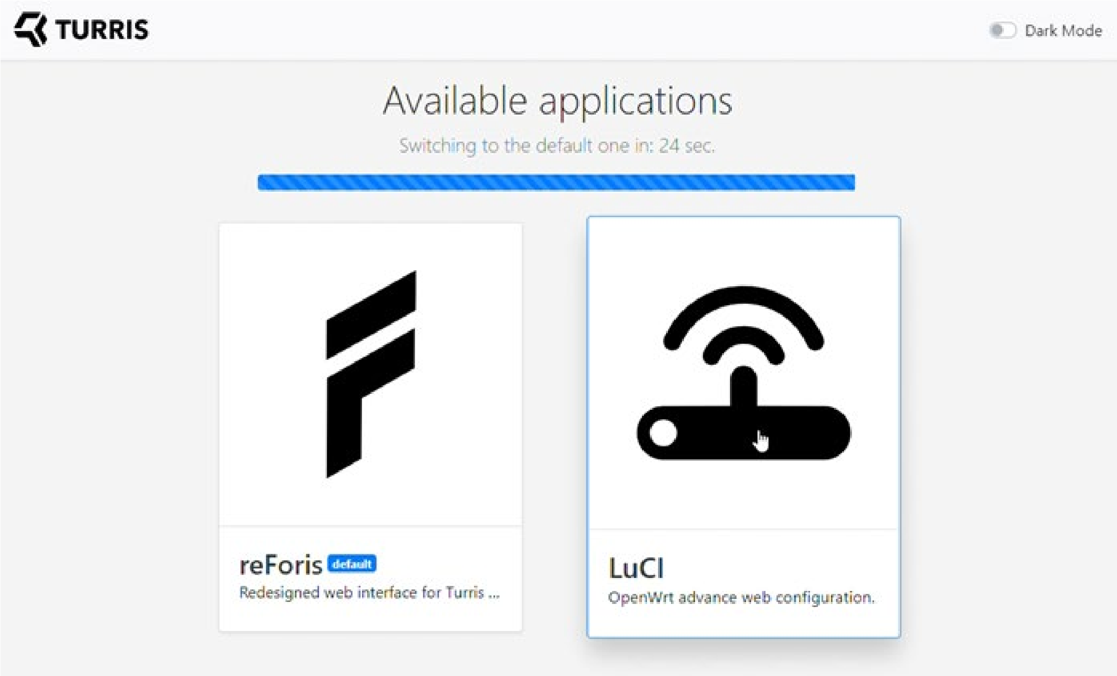
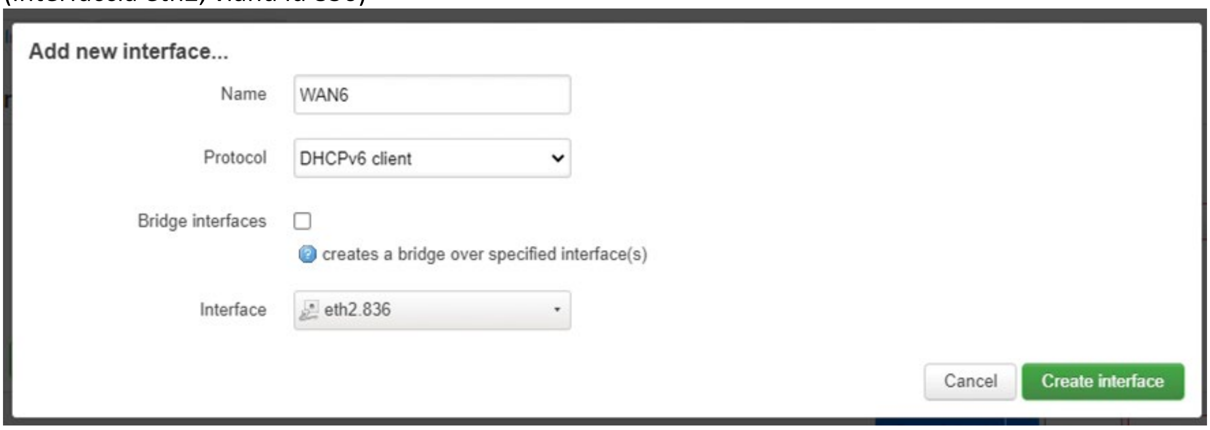
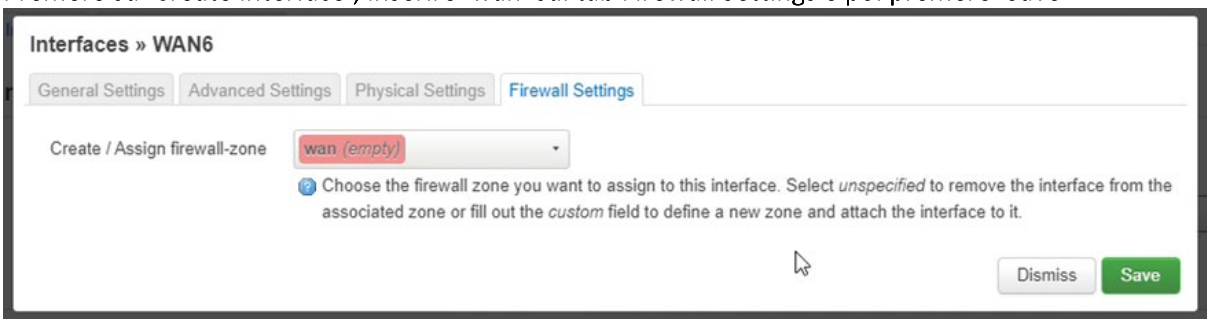
1. Aprire il browser, digitare 192.168.1.1 e selezionare l'interfaccia LuCI



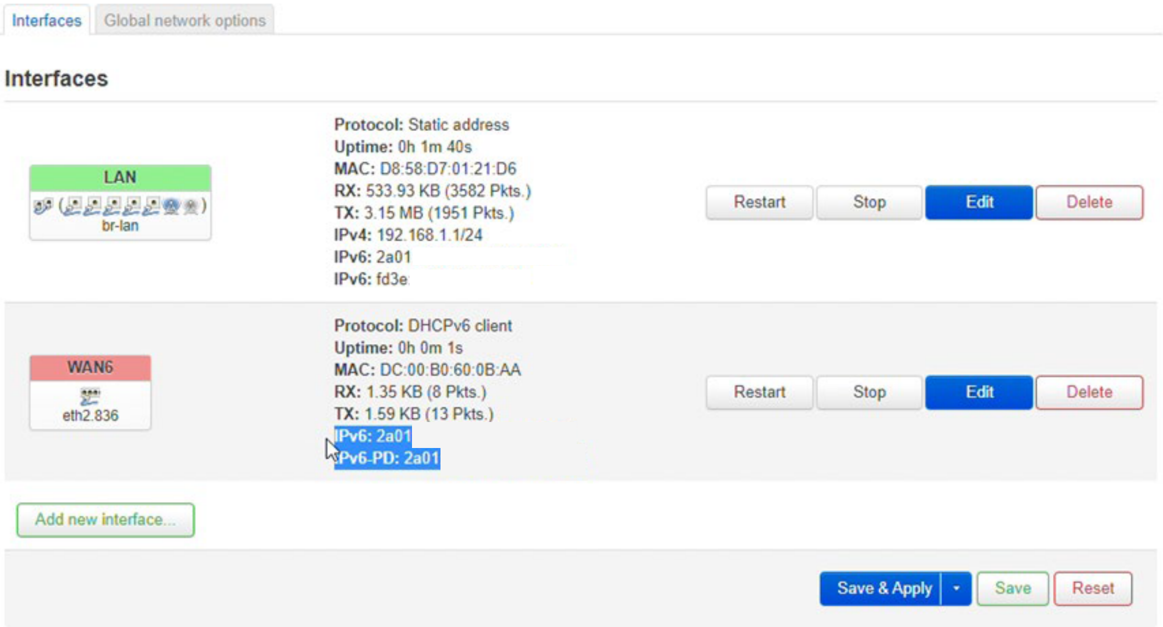
2. Inserire i dati di Login e poi andare in 'Network' → 'Interfaces' e poi premere su 'Add new interface…'
   Inserire name 'WAN6', Protocol 'DHCPv6 client' e creare una custom interface 'eth2.836' (interfaccia eth2, vland id 836)



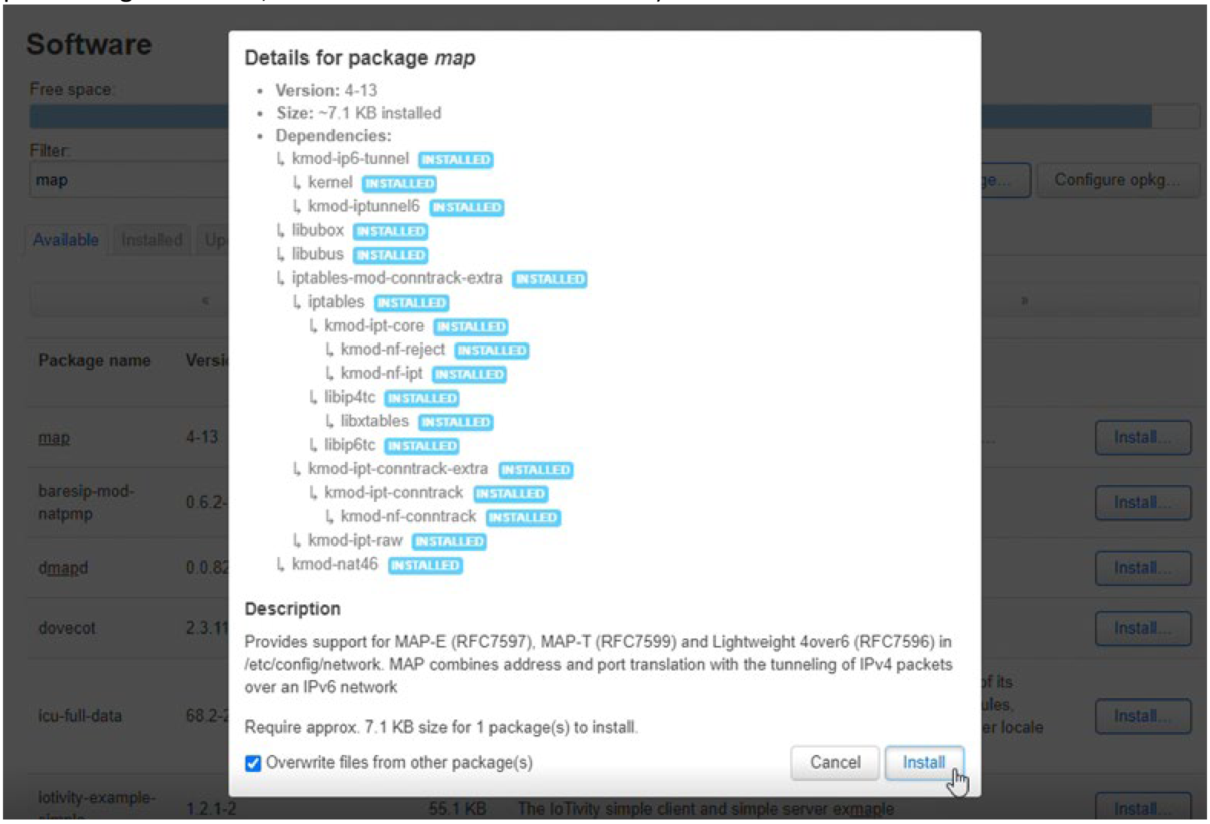
3. Premere su 'Create interface', inserire 'wan' sul tab Firewall Settings e poi premere 'Save'



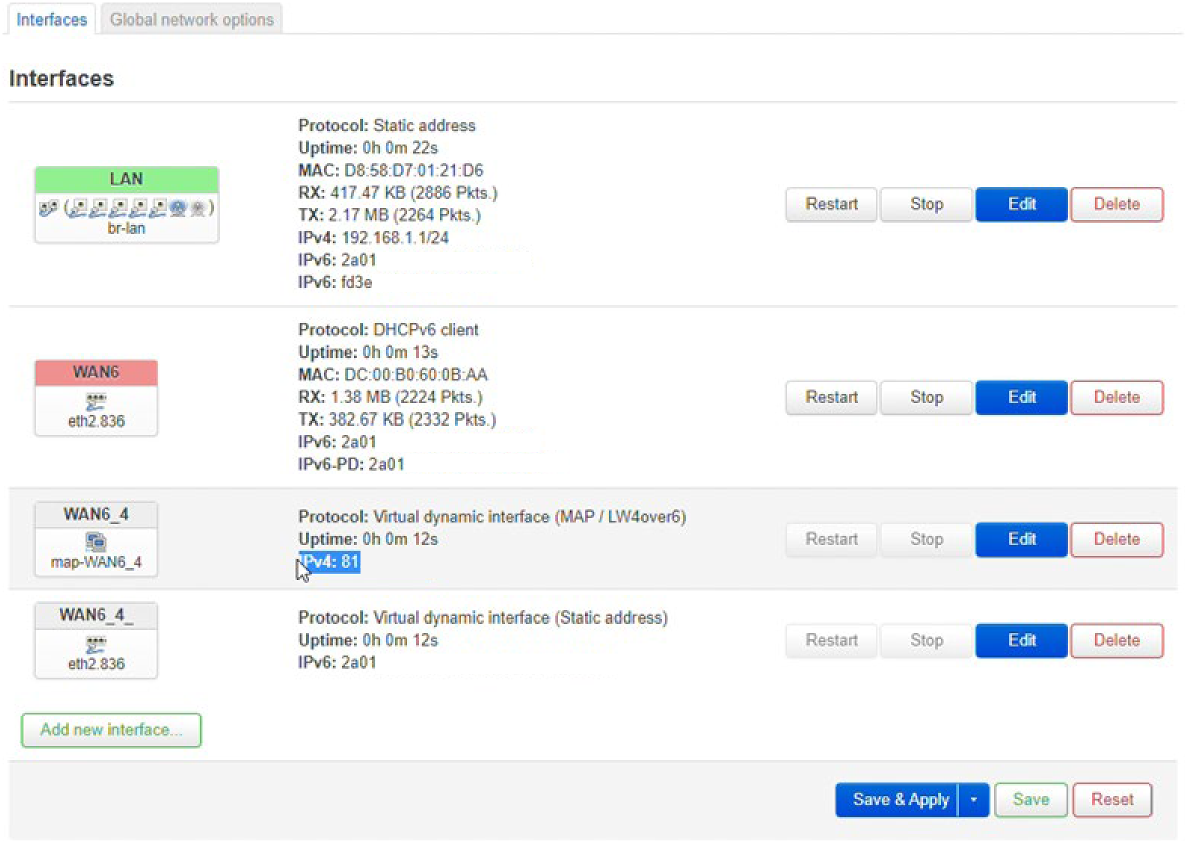
4. Premere su 'Save & Apply' ed aggiornare la pagina, a questo punto vedremo l'IPv6 assegnato e potremo navigare su Internet per scaricare la libreria MAP



5. Per scaricare il modulo MAP andare in 'System' → 'Software' e poi premere sul bottone verde 'Update lists…'
A questo punto cerchiamo il modulo MAP e installiamolo premendo 'Install' (se ci sono pacchetti già installati, inserire il check su 'Overwrite…')



6. Ora riavviare il router andando su ‘System’ → ‘Reboot’ e cliccando ‘Perform reboot’
Una volta riavviato, possiamo ritornare su ‘Network’ → ‘Interface’ e vedere l’IPv4 assegnato



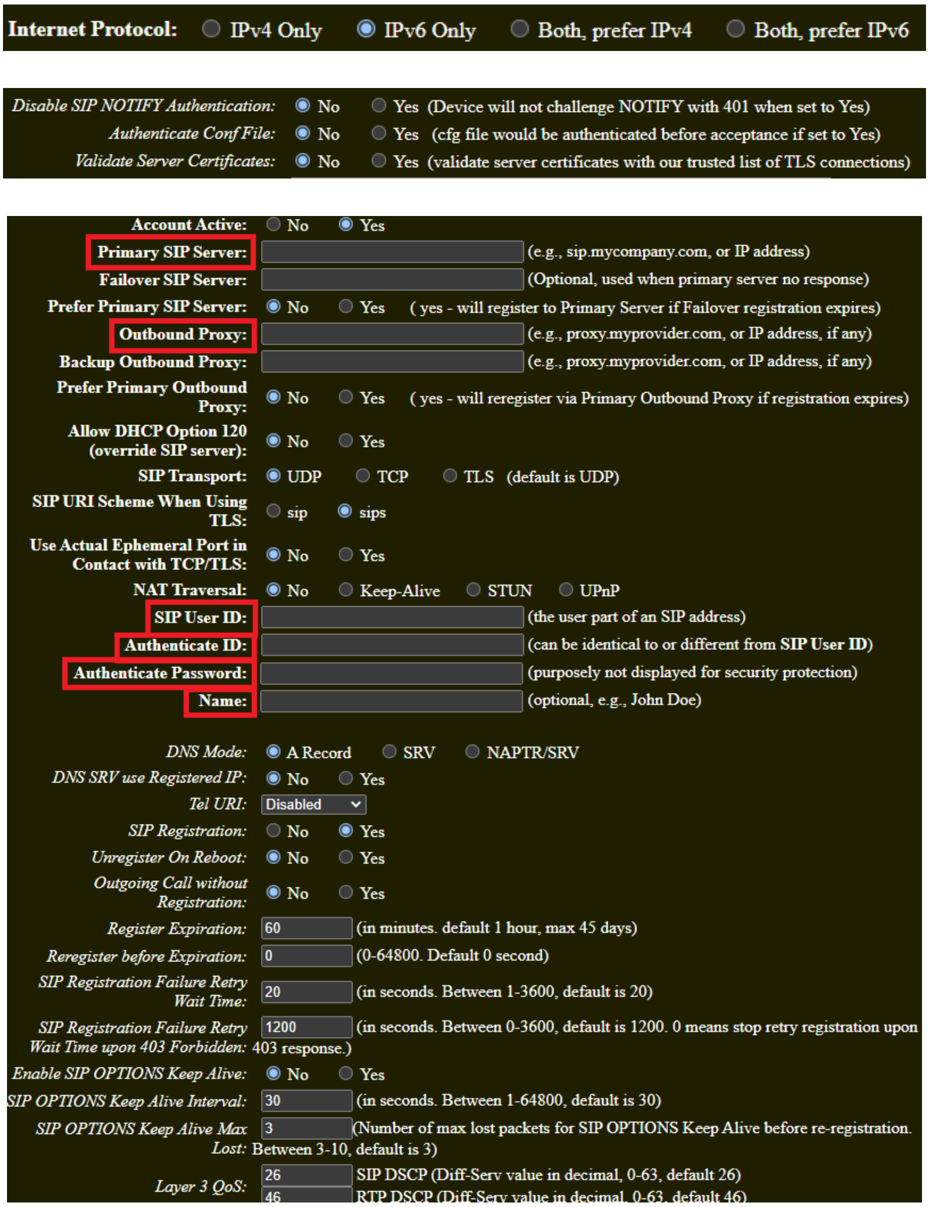
7. Accedere nella propria area personale nella sezione ‘I miei dispositivi /Informazioni Net Neutrality‘ ed aggiungere il MAC del proprio apparato

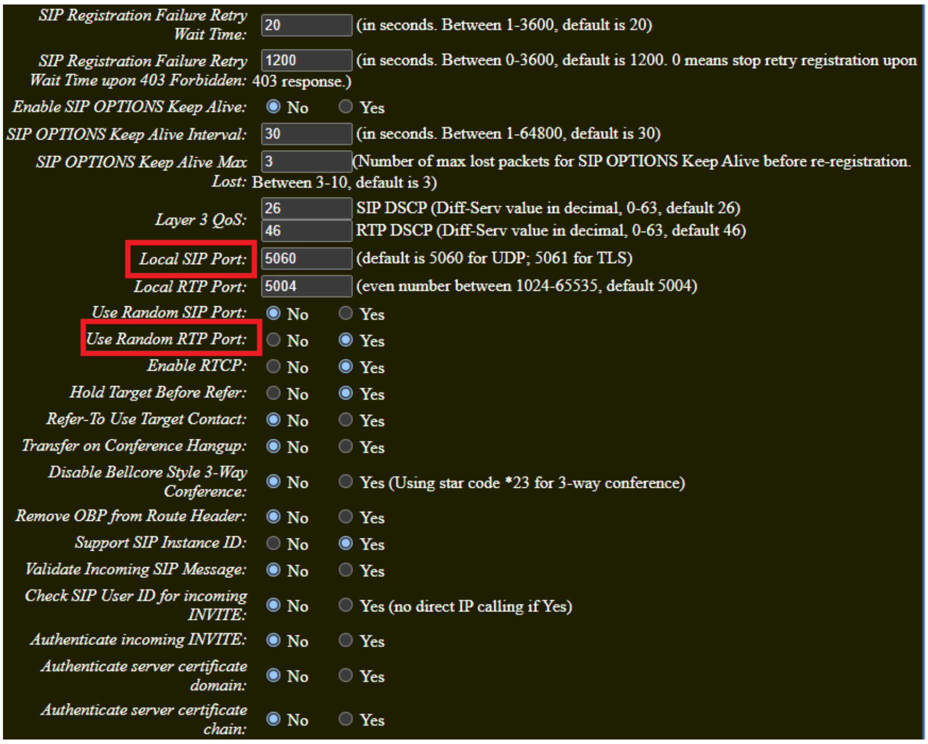
CONFIGURAZIONE VOIP – ESEMPIO CON GRANDSTREAM H813

Di seguito i dati necessari per la configurazione del dispositivo e le relative schermate.

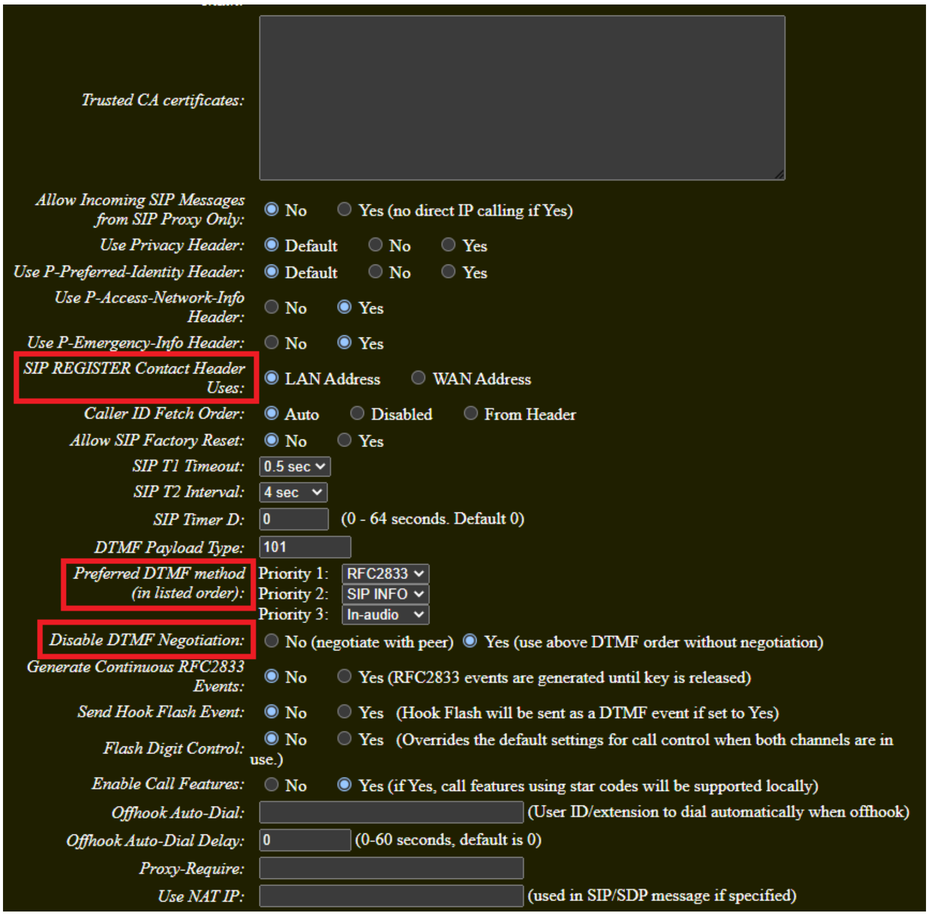
- **SIP USERNAME:** Dato disponibile in area personale
- **SIP PASSWORD:** Dato disponibile in area personale
- **SIP DOMAIN:** voip.iliad.it
- **SIP OUTBOUND PROXY:** Dato disponibile in area personale
- **SIP PORT:** 5060
- **SIP PROTOCOL:** UDP

https://www.grandstream.com/hubfs/Product_Documentation/HT813_User_Guide.pdf



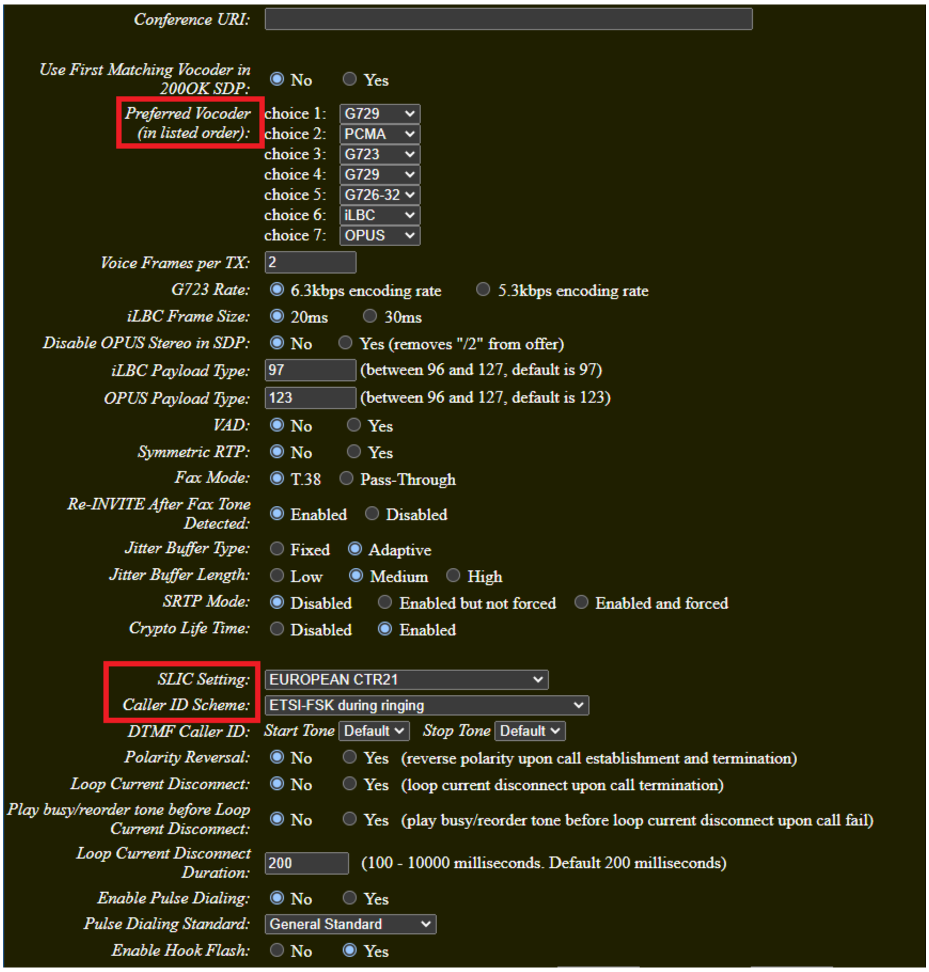


SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP Registration Failure Retry Wait Time upon 403 Forbidden:	1200 (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)
Enable SIP OPTIONS Keep Alive:	No / Yes
SIP OPTIONS Keep Alive Interval:	30 (in seconds. Between 1-64800, default is 30)
SIP OPTIONS Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)
Local SIP Port:	5060 (default is 5060 for UDP; 5061 for TLS)
Local RTP Port:	5004 (even number between 1024-65535, default 5004)
Use Random SIP Port:	No / Yes
Use Random RTP Port:	No / Yes
Enable RTCP:	No / Yes
Hold Target Before Refer:	No / Yes
Refer-To Use Target Contact:	No / Yes
Transfer on Conference Hangup:	No / Yes
Disable Bellcore Style 3-Way Conference:	No / Yes (Using star code *23 for 3-way conference)
Remove OBP from Route Header:	No / Yes
Support SIP Instance ID:	No / Yes
Validate Incoming SIP Message:	No / Yes
Check SIP User ID for incoming INVITE:	No / Yes (no direct IP calling if Yes)
Authenticate incoming INVITE:	No / Yes
Authenticate server certificate domain:	No / Yes
Authenticate server certificate chain:	No / Yes



Trusted CA certificates:	
Allow Incoming SIP Messages from SIP Proxy Only:	No / Yes (no direct IP calling if Yes)
Use Privacy Header:	Default / No / Yes
Use P-Preferred-Identity Header:	Default / No / Yes
Use P-Access-Network-Info Header:	No / Yes
Use P-Emergency-Info Header:	No / Yes
SIP REGISTER Contact Header Uses:	LAN Address / WAN Address
Caller ID Fetch Order:	Auto / Disabled / From Header
Allow SIP Factory Reset:	No / 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 (in listed order):	Priority 1: RFC2833 Priority 2: SIP INFO Priority 3: In-audio
Disable DTMF Negotiation:	No (negotiate with peer) / Yes (use above DTMF order without negotiation)
Generate Continuous RFC2833 Events:	No / Yes (RFC2833 events are generated until key is released)
Send Hook Flash Event:	No / Yes (Hook Flash will be sent as a DTMF event if set to Yes)
Flash Digit Control:	No / Yes (Overrides the default settings for call control when both channels are in use.)
Enable Call Features:	No / Yes (if Yes, call features using star codes will be supported locally)
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
Offhook Auto-Dial Delay:	0 (0-60 seconds, default is 0)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)

SIP User-Agent:	
SIP User-Agent Postfix:	
Disable Call-Waiting:	No Yes
Disable Call-Waiting Caller ID:	No Yes
Disable Call-Waiting Tone:	No Yes
Disable Connected Line ID:	No Yes
Disable Receiver Offhook Tone:	No Yes (ROH tone will not be played after offhook for 60 seconds)
Disable Reminder Ring for On-Hold Call:	No Yes
Disable Visual MWI:	No Yes
Do Not Escape '#' as %23 in SIP URI:	No Yes
Disable Multiple m line in SDP:	No Yes
Ring Timeout:	60 (0-300, default is 60 seconds, 0 means no timeout)
Delayed Call Forward Wait Time:	20 (Allowed range 1-120, in seconds.)
No Key Entry Timeout:	4 (1-15, default is 4 seconds)
Early Dial:	No 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:	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 Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous:	No Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection:	No Yes
Special Feature:	Standard
Enable Session Timer:	No Yes
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:	UAC UAS Omit (Recommended)
UAS Specify Refresher:	UAC UAS (When UAC did not specify refresher tag)
Force INVITE:	No Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel:	No Yes
Add Auth Header On Initial REGISTER:	No Yes


Conference URI:
Use First Matching Vocoder in 200OK SDP: No Yes
Preferred Vocoder (in listed order):
choice 1: G729
choice 2: PCMA
choice 3: G723
choice 4: G729
choice 5: G726-32
choice 6: iLBC
choice 7: OPUS
Voice Frames per TX: 2
G723 Rate: 6.3kbps encoding rate 5.3kbps encoding rate
iLBC Frame Size: 20ms 30ms
Disable OPUS Stereo in SDP: No Yes (removes "/2" from offer)
iLBC Payload Type: 97 (between 96 and 127, default is 97)
OPUS Payload Type: 123 (between 96 and 127, default is 123)
VAD: No Yes
Symmetric RTP: No Yes
Fax Mode: T.38 Pass-Through
Re-INVITE After Fax Tone Detected: Enabled Disabled
Jitter Buffer Type: Fixed Adaptive
Jitter Buffer Length: Low Medium High
SRTP Mode: Disabled Enabled but not forced Enabled and forced
Crypto Life Time: Disabled Enabled
SLIC Setting: EUROPEAN CTR21
Caller ID Scheme: ETSI-FSK during ringing
DTMF Caller ID: Start Tone Default Stop Tone Default
Polarity Reversal: No Yes (reverse polarity upon call establishment and termination)
Loop Current Disconnect: No 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 Yes
Pulse Dialing Standard: General Standard
Enable Hook Flash: No Yes

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