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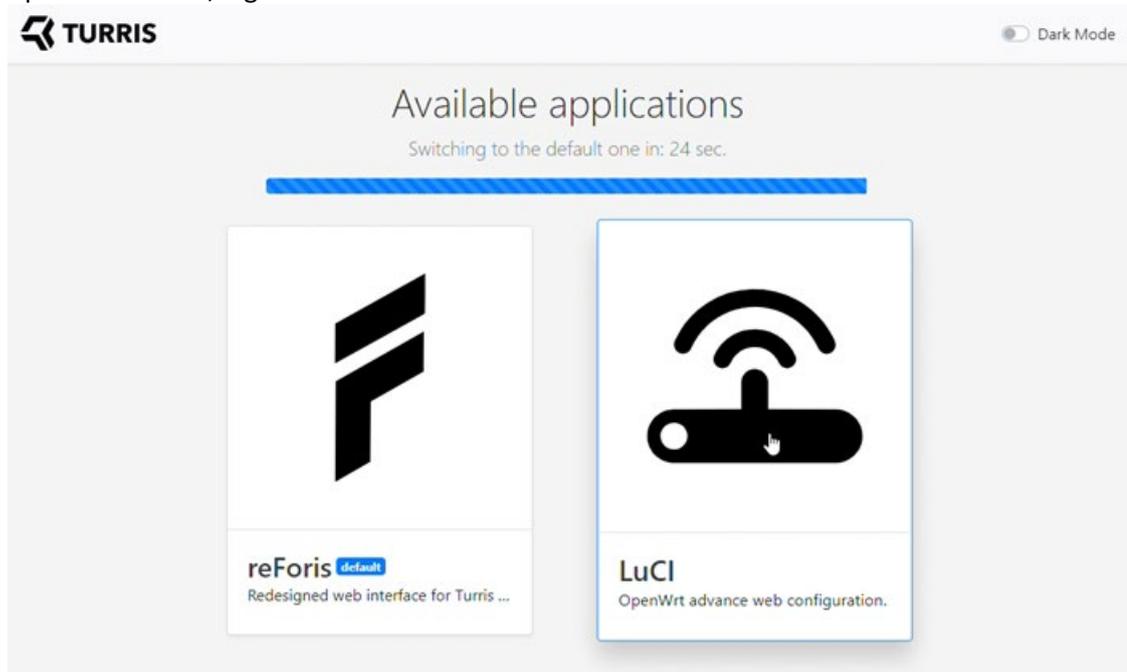
CONFIGURAZIONE TURRIS



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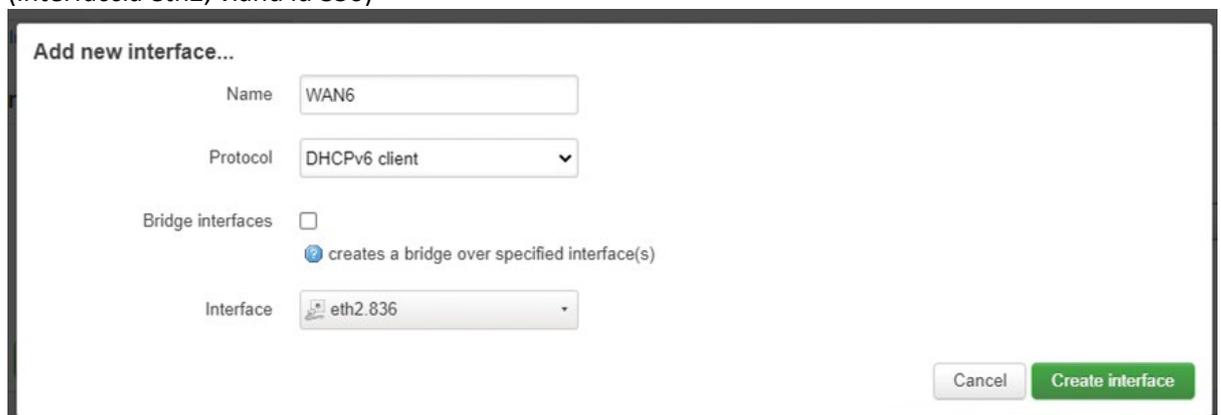
CONFIGURAZIONE TURRIS

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)

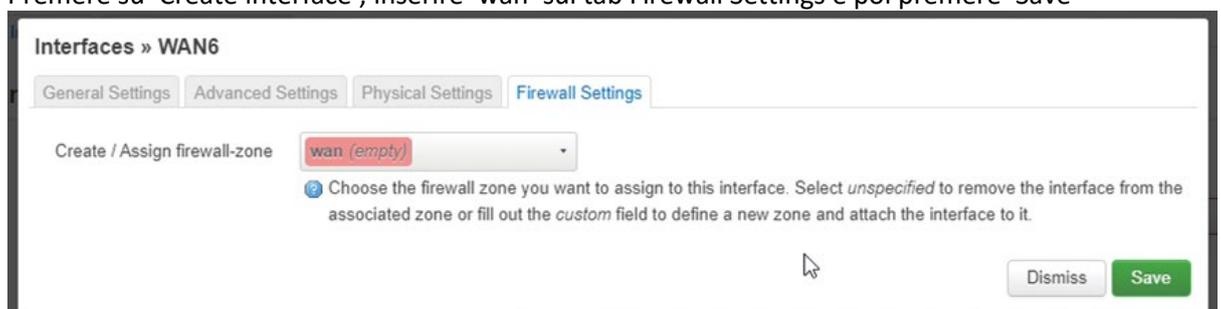


The screenshot shows the 'Add new interface...' form. It has the following fields and options:

- Name: WAN6
- Protocol: DHCPv6 client
- Bridge interfaces: (with a tooltip: creates a bridge over specified interface(s))
- Interface: eth2.836

At the bottom right, there are 'Cancel' and 'Create interface' buttons.

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



The screenshot shows the 'Interfaces » WAN6' page with the 'Firewall Settings' tab selected. It has the following elements:

- Navigation tabs: General Settings, Advanced Settings, Physical Settings, Firewall Settings
- Create / Assign firewall-zone: wan (empty)
- Help text: Choose the firewall zone you want to assign to this interface. Select *unspecified* to remove the interface from the associated zone or fill out the *custom* field to define a new zone and attach the interface to it.

At the bottom right, there are 'Dismiss' and 'Save' buttons.

- 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

The screenshot shows the Mikrotik WinBox 'Interfaces' configuration page. The 'LAN' interface (br-lan) is highlighted in green, and the 'WAN6' interface (eth2.836) is highlighted in red. The WAN6 interface details show it is configured as a DHCPv6 client with IPv6 address 2a01:fd3e:2a01:2a01::1. Buttons for 'Restart', 'Stop', 'Edit', and 'Delete' are visible for both interfaces. At the bottom, there are buttons for 'Save & Apply', 'Save', and 'Reset'.

- 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...')

The screenshot shows the Mikrotik WinBox 'Software' page. The 'map' package is selected, and its details are shown in a modal window. The package is listed as 'map' with version '4-13'. The details window shows a list of dependencies, all of which are marked as 'INSTALLED'. The description states: 'Provides support for MAP-E (RFC7597), MAP-T (RFC7599) and Lightweight 4over6 (RFC7596) in /etc/config/network. MAP combines address and port translation with the tunneling of IPv4 packets over an IPv6 network.' The size is approximately 7.1 KB. The 'Overwrite files from other package(s)' checkbox is checked. Buttons for 'Cancel' and 'Install' are visible at the bottom of the details window.

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

Interfaces Global network options

Interfaces

LAN br-lan	Protocol: Static address Uptime: 0h 0m 22s MAC: D8:58:D7:01:21:D6 RX: 417.47 KB (2886 Pkts.) TX: 2.17 MB (2264 Pkts.) IPv4: 192.168.1.1/24 IPv6: 2a01 IPv6: fd3e	Restart Stop Edit Delete
WAN6 eth2.836	Protocol: DHCPv6 client Uptime: 0h 0m 13s MAC: DC:00:B0:60:0B:AA RX: 1.38 MB (2224 Pkts.) TX: 382.67 KB (2332 Pkts.) IPv6: 2a01 IPv6-PD: 2a01	Restart Stop Edit Delete
WAN6_4 map-WAN6_4	Protocol: Virtual dynamic interface (MAP / LW4over6) Uptime: 0h 0m 12s IPv4: 81	Restart Stop Edit Delete
WAN6_4 eth2.836	Protocol: Virtual dynamic interface (Static address) Uptime: 0h 0m 12s IPv6: 2a01	Restart Stop Edit Delete

Add new interface...

Save & Apply Save Reset

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

Internet Protocol: IPv4 Only IPv6 Only Both, prefer IPv4 Both, prefer IPv6

Disable SIP NOTIFY Authentication: No Yes (Device will not challenge NOTIFY with 401 when set to Yes)

Authenticate Conf File: No Yes (cfg file would be authenticated before acceptance if set to Yes)

Validate Server Certificates: No Yes (validate server certificates with our trusted list of TLS connections)

Account Active: No Yes

Primary SIP Server: (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: (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)

Allow DHCP Option 120 (override SIP server): No Yes

SIP Transport: UDP TCP TLS (default is UDP)

SIP URI Scheme When Using TLS: sip sips

Use Actual Ephemeral Port in Contact with TCP/TLS: No Yes

NAT Traversal: No Keep-Alive STUN UPnP

SIP User ID: (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 Record SRV NAPTR/SRV

DNS SRV use Registered IP: No Yes

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Register Expiration: (in minutes, default 1 hour, max 45 days)

Reregister before Expiration: (0-64800, Default 0 second)

SIP Registration Failure Retry Wait Time: (in seconds, Between 1-3600, default is 20)

SIP Registration Failure Retry Wait Time upon 403 Forbidden: (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: (in seconds, Between 1-64800, default is 30)

SIP OPTIONS Keep Alive Max Lost: (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)

Layer 3 QoS: SIP DSCP (Diff-Serv value in decimal, 0-63, default 26)

RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)

SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)

SIP Registration Failure Retry Wait Time upon 403 Forbidden: (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)

Enable SIP OPTIONS Keep Alive: No Yes

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Layer 3 QoS: SIP DSCP (Diff-Serv value in decimal, 0-63, default 26)
 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)

Local SIP Port: (default is 5060 for UDP; 5061 for TLS)

Local RTP Port: (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

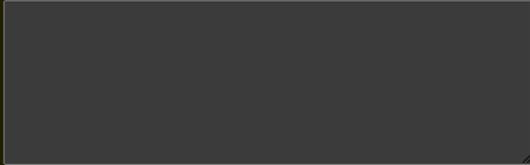
Validate Incoming SIP Message: No Yes

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:

SIP T2 Interval:

SIP Timer D: (0 - 64 seconds. Default 0)

DTMF Payload Type:

Preferred DTMF method (in listed order):
 Priority 1:
 Priority 2:
 Priority 3:

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: (0-60 seconds, default is 0)

Proxy-Require:

Use NAT IP: (used in SIP/SDP message if specified)

SIP User-Agent:	<input type="text"/>
SIP User-Agent Postfix:	<input type="text"/>
Disable Call-Waiting:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Call-Waiting Caller ID:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Call-Waiting Tone:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Connected Line ID:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Receiver Offhook Tone:	<input checked="" type="radio"/> No <input type="radio"/> Yes (ROH tone will not be played after offhook for 60 seconds)
Disable Reminder Ring for On-Hold Call:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Visual MWI:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Do Not Escape '#' as %23 in SIP URI:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Multiple m line in SDP:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ring Timeout:	<input type="text" value="60"/> (0-300, default is 60 seconds, 0 means no timeout)
Delayed Call Forward Wait Time:	<input type="text" value="20"/> (Allowed range 1-120, in seconds.)
No Key Entry Timeout:	<input type="text" value="4"/> (1-15, default is 4 seconds)
Early Dial:	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	<input type="text"/> (this prefix string is added to each dialed number)
Use # as Dial Key:	<input type="radio"/> No <input checked="" type="radio"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan:	<input type="text" value="{x+ v*x+ *x+ *xx*x+}"/>
SUBSCRIBE for MWI:	<input checked="" type="radio"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="radio"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous:	<input checked="" type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Special Feature:	Standard ▾
Enable Session Timer:	<input type="radio"/> No <input checked="" type="radio"/> Yes
Session Expiration:	<input type="text" value="180"/> (90-64800, default 180 seconds)
Min-SE:	<input type="text" value="90"/> (90-64800, default 90 seconds)
Caller Request Timer:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Request for timer when making outbound calls)
Callee Request Timer:	<input checked="" type="radio"/> No <input type="radio"/> Yes (When caller supports timer but did not request one)
Force Timer:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Use timer even when remote party does not support)
UAC Specify Refresher:	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)
UAS Specify Refresher:	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)
Force INVITE:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Add Auth Header On Initial REGISTER:	<input checked="" type="radio"/> No <input type="radio"/> Yes

Conference URI:

Use First Matching Vocoder in 200OK SDP: No Yes

Preferred Vocoder (in listed order):

choice 1:

choice 2:

choice 3:

choice 4:

choice 5:

choice 6:

choice 7:

Voice Frames per TX:

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: (between 96 and 127, default is 97)

OPUS Payload Type: (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:

Caller ID Scheme:

DTMF Caller ID: Start Tone Stop Tone

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: (100 - 10000 milliseconds. Default 200 milliseconds)

Enable Pulse Dialing: No Yes

Pulse Dialing Standard:

Enable Hook Flash: No Yes

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