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CONFIGURAZIONE MIKROTIK - GUI

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CONNESSIONE ALL'APPARATO

Effettuare il download di Winbox dal seguente link: https://mikrotik.com/download
Note: Verificare compatibilità con sistema operativo utilizzato. Se compatibile, procedere con gli step seguenti.
Se non compatibile, seguire la guida CONFIGURAZIONE MIKROTIK – CLI

1. Collegare il router Mikrotik e il proprio PC tramite cavo RJ-45. Di default, tutte le porte del router sono attive.

2. Aprire Winbox sul device utilizzato per effettuare la configurazione e verificare che nella sezione neighbors venga rilevato il router Mikrotik
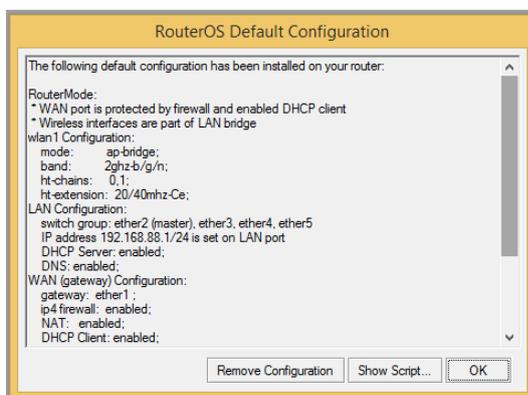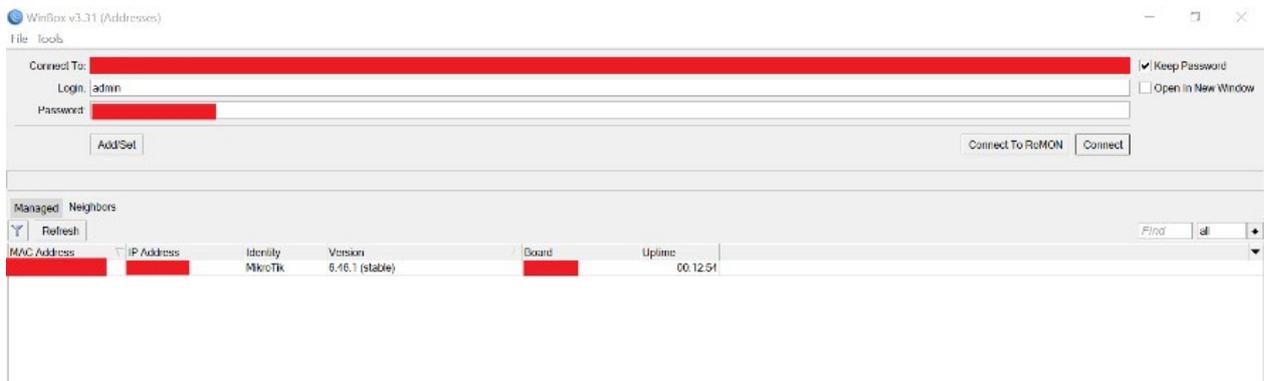
CONFIGURAZIONE INTERFACCE FISICHE/VIRTUALI E WIRELESS

6. Partendo dal punto 5:
   - Selezionare System → Packages → Selezionare ipv6 → Enable
   - Selezionare System → Reboot: Yes

8. Partendo dal punto 5:
   - Selezionare Wireless → doppio click su default → Mode: dynamic keys → Authentication Types: WPA PSK, WPA2 PSK → Unicast Ciphers: aes ccm,tkip → Group Ciphers: aes ccm,tkip
   Note: Nelle sezioni succcessive "WPA Pre-Shared Key" e "WPA2 Pre-Shared Key" configurare la password del WIFI associata all'SSID scelto nel punto 6

CONFIGURAZIONE BRIDGE

9. Partendo dal punto 5:
   - Selezionare Bridge → Ports → + → Interface: sfp-sfpplus1 → Bridge: WAN → OK
   - Selezionare Bridge → Ports → + → Interface: ether1 → Bridge: LAN → OK
   - Selezionare Bridge → Ports → + → Interface: ether2 → Bridge: LAN → OK
   - Selezionare Bridge → Ports → + → Interface: ether3 → Bridge: LAN → OK
   […] fino alla porta ether9
   - Selezionare Bridge → Ports → + → Interface: wlan1 → Bridge: LAN → OK
   - Selezionare Bridge → Ports → + → Interface: wlan2 → Bridge: LAN → OK
   - Selezionare Bridge → Ports → + → Interface: ether10 → Bridge: LAN-VOIP → OK

CONFIGURAZIONE IPv4 ADDRESSING

10. Partendo dal punto 5:
    - Selezionare IP → Addresses → + → Address: Dato disponibile in area personale → Interface: ipipv6-tunnel1 →OK
    - Selezionare IP → Addresses → + → Address: 192.168.1.1/24 → Interface: LAN → OK
    Note: L'IP utilizzato è d'esempio, è possibile utilizzare qualsiasi altra classe IP privata riportata nella documentazione RFC 1597 e 1918
    - Selezionare IP → Addresses → + → Address: 192.168.2.1/24 → Interface: LAN-VOIP→ OK
    - Selezionare IP → Pool → + →Name: DHCP-LAN → Addresses: 192.168.1.2-192.168.1.254 → OK
    Note: Gli IP utilizzati sono d'esempio, è possibile utilizzare qualsiasi altra classe IP riportata nella documentazione RFC 1597 e 1918.
    - Selezionare IP → DHCP Server → + →Name: DHCP-LAN → Interface: LAN → Address Pool: DHCP-LAN → OK
    - Selezionare IP → DHCP Server → Networks → + → Address: 192.168.1.0/24 → Gateway: 192.168.1.1 → DNS Server: 8.8.8.8,8.8.4.4 → OK
    Note: Verificare che nella sezione Leases il dispositivo/dispositivi collegati abbiano ricevuto un IP dal DHCP server (in questo caso la Mikrotik)
    - Selezionare IP → Pool → + →Name: DHCP-VOIP-LAN→ Addresses: 192.168.2.2-192.168.2.254 → OK
    Note: Gli IP utilizzati sono d'esempio, è possibile utilizzare qualsiasi altra classe IP riportata nella documentazione RFC 1597 e 1918. In questa sezione è fondamentale configurare un range di IP disponibili.
    - Selezionare IP → DHCP Server → + →Name: DHCP-VOIP-LAN → Interface: LAN-VOIP → Address Pool: DHCP-VOIP-LAN → OK
    - Selezionare IP → DHCP Server → Networks → + → Address: 192.168.2.0/24 → Gateway: 192.168.2.1 → DNS Server: 8.8.8.8,8.8.4.4 → OK
    Note: Verificare che nella sezione Leases il dispositivo/dispositivi collegati abbiano ricevuto un IP dal DHCP server (in questo caso la Mikrotik)
    - Selezionare IP → Firewall → NAT → + → Chain: srcnat → Src. Address: 192.168.1.0/24 → Action → Action: masquerade → OK
    - Selezionare IP → Routes → + → Dst. Address: 0.0.0.0/0 → Gateway: ipipv6-tunnel1 → OK

CONFIGURAZIONE IPv6 ADDRESSING

11. Partendo dal punto 5:
    - Selezionare IPv6 → Addresses → + → Addresses: Dato disponibile in area personale → Interface: ipipv6-tunnel1 → OK
    - Selezionare IPv6 → DHCP Client → + → DHCP → Interface: WAN:836 → Request: Selezionare address,prefix → Pool Name: ipv6-pool → Selezionare "Use Peer DNS, Rapid Commit, Add default Route → OK → Click sulla voce "Release"
    - Selezionare IPv6 → Addresses → + → Addresses: ::192:168:1:1/64 → From Pool: : ipv6-pool → Interface: LAN → Selezionare Advertise → OK
    - Selezionare IPv6 → Addresses → + → Addresses: ::192:168:2:1/64 → From Pool: : ipv6-pool → Interface: LAN-VOIP → Selezionare Advertise → OK
    - Selezionare IPv6 → Routes → + → Dst. Address: Dato disponibile in area personale → Gateway: WAN → OK

PERSONALIZZARE CREDENZIALI DI ACCESSO ALL'APPARATO

Come best practices una volta effettuate le configurazioni, si consiglia dalla schermata di winbox (punto 5):

- New Terminal → copiare e incollare
 /user add name=myname password=mypassword group=full
 /user remove admin

Note: "Myname" e "mypassword" sono valori di default che devono essere cambiati con le credenziali che si vogliono utilizzare per collegarsi all'apparato.

- New Terminal → copiare e incollare
/ip service
set telnet disabled=yes
set ftp disabled=yes
set www disabled=yes
set api disabled=yes
set api-ssl disabled=yes

Per maggiori dettagli

- https://wiki.mikrotik.com/wiki/Manual:Securing_Your_Router
- https://wiki.mikrotik.com/wiki/Main_Page

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

Enable Hook Flash:	○ 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:	*TX* 0dB default ▾ *RX* -6dB default ▾	
Disable Line Echo Canceller (LEC):	● No ○ 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 ○ Yes	
RFC2833 Events Count:	8 (between 2 and 10, default is 8)	
RFC2833 End Events Count:	3 (between 2 and 10, default is 3)	

	Ring Tone 1 ▾	used if incoming caller ID is
Distinctive Ring Tone:	Ring Tone 1 ▾	used if incoming caller ID is
	Ring Tone 1 ▾	used if incoming caller ID is

Ring Tones (Syntax: c=on1/off1-on2/off2-on3/off3;)

Ring Tone 1:	c=2000/4000;
Ring Tone 2:	c=2000/4000;
Ring Tone 3:	c=2000/4000;
Ring Tone 4:	c=2000/4000;
Ring Tone 5:	c=2000/4000;
Ring Tone 6:	c=2000/4000;
Ring Tone 7:	c=2000/4000;
Ring Tone 8:	c=2000/4000;
Ring Tone 9:	c=2000/4000;
Ring Tone 10:	c=2000/4000;

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