

# iliad

**CONFIGURAZIONE MIKROTIK - GUI**



[iliad.it](http://iliad.it)

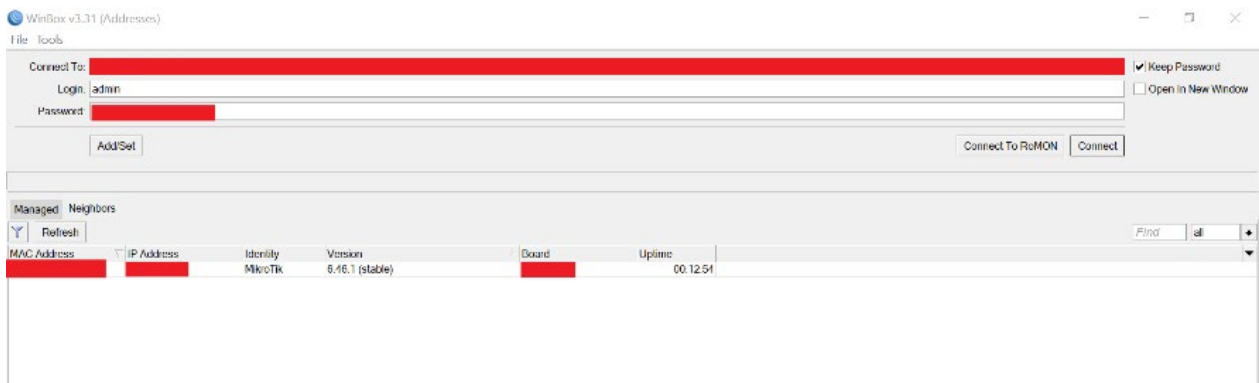
## CONFIGURAZIONE MIKROTIK - GUI

### CONNESSIONE ALL'APPARATO

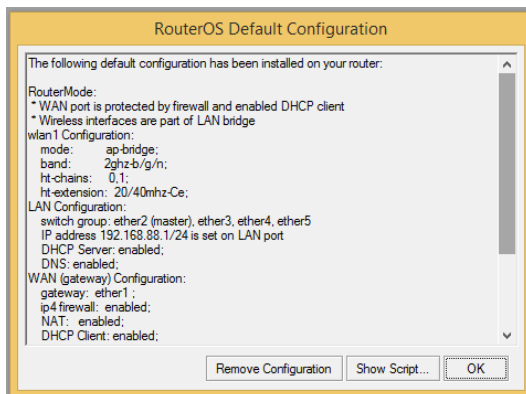
Effettuare il download di Winbox dal seguente link: <https://mikrotik.com/download>

Note: Verificare compatibilità con sistema operativo utilizzato. Se compatibile, procedere con gli step seguenti.

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
3. Effettuare il login con i parametri di default – Login: admin (no password)



4. Rimuovere la “default config” che si presenta al primo avvio come da esempio che segue e attendere il reboot. Al termine del riavvio, effettuare nuovamente il login seguendo i punti 2 e 3.



5. La schermata di winbox dovrà presentarsi come segue:



## CONFIGURAZIONE INTERFACCE FISICHE/VIRTUALI E WIRELESS

6. Partendo dal punto 5:
  - Selezionare System → Packages → Selezionare ipv6 → Enable
  - Selezionare System → Reboot: Yes
  
7. Partendo dal punto 5:
  - Selezionare Interfaces → + → Bridge → Name : WAN → OK
  - Selezionare Interfaces → + → Bridge → Name : LAN → OK
  - Selezionare Interfaces → + → Bridge → Name : LAN-VOIP → OK
  - Selezionare Interfaces → + → VLAN → Name: WAN:836 → VLAN ID: 836 → Interface: WAN
  - Selezionare Interfaces → + → IPIPv6 Tunnel → Name: ipv6-tunnel1 → Local Address: **Dato disponibile in area personale** → Remote Address: **Dato disponibile in area personale** → OK
  - Selezionare Interfaces → wlan1 o wlan2 → Wireless → Mode: AP bridge

*Note: Avendo la possibilità di modificare i parametri in base alle proprie esigenze, un esempio di configurazione potrebbe essere come segue:*

The screenshot shows the configuration page for Interface <wlan1> in WinBox. The 'Wireless' tab is active, and several settings are highlighted with yellow boxes:

- Mode: ap bridge
- Band: 5GHz-A/N/AC
- Channel Width: 20/40/80MHz Ceee
- Frequency: 5180
- SSID: Home
- Country: italy
- Installation: indoor
- WMM Support: disabled
- Bridge Mode: enabled
- VLAN Mode: no tag
- VLAN ID: 1
- Default AP Tx Limit: (empty)
- Default Client Tx Limit: (empty)
- Default Authenticate:
- Default Forward:
- Hide SSID:
- Multicast Helper: default
- Multicast Buffering:
- Keepalive Frames:

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 successive "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/Manual:Securing_Your_Router)

- [https://wiki.mikrotik.com/wiki/Main\\_Page](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](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)

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**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

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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:  (0-300, default is 60 seconds, 0 means no timeout)

Delayed Call Forward Wait Time:  (Allowed range 1-120, in seconds.)

No Key Entry Timeout:  (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:  (this prefix string is added to each dialed number)

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:

Enable Session Timer:  No  Yes

Session Expiration:  (90-64800, default 180 seconds)

Min-SE:  (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:   
 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:

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



*Enable Hook Flash:*  No  Yes

*Hook Flash Timing:* In 40-2000 milliseconds range, minimum:  maximum:

*On Hook Timing:*  (In 40-2000 milliseconds range, default is 400)

*Gain:* TX  RX

*Disable Line Echo Canceller (LEC):*  No  Yes

*Disable Network Echo Suppressor:*  No  Yes

*Outgoing Call Duration Limit:*  (0-180 minutes, default is 0 (No Limit) )

*Enable High Ring Power:*  No  Yes

*RFC2833 Events Count:*  (between 2 and 10, default is 8)

*RFC2833 End Events Count:*  (between 2 and 10, default is 3)

*Distinctive Ring Tone:*

<input type="text" value="Ring Tone 1"/>	used if incoming caller ID is	<input type="text"/>
<input type="text" value="Ring Tone 1"/>	used if incoming caller ID is	<input type="text"/>
<input type="text" value="Ring Tone 1"/>	used if incoming caller ID is	<input type="text"/>

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

<i>Ring Tone 1:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 2:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 3:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 4:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 5:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 6:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 7:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 8:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 9:</i>	<input type="text" value="c=2000/4000;"/>
<i>Ring Tone 10:</i>	<input type="text" value="c=2000/4000;"/>

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