KEY FEATURES

- Make and receive VoIP calls from a single telephone handset
- Landline backup in the event of a failure with your VoIP Service
- 1 FXS, 1 FXO port and 1 LAN port
- Simple web-based setup and configuration
- Supports advanced Call services – Caller ID, Call On-Hold, Call Forward
- Call Waiting and Transfer – service provider dependent*

*Note: The availability of some listed call features are dependent on the service supported by your VoIP service provider. Please consult them for further information.

<table>
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<tr>
<th>Model Code/ Part Number</th>
<th>Description</th>
<th>APN Code</th>
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<tbody>
<tr>
<td>V210P</td>
<td>NetComm Express™ Series VoIP Adapter</td>
<td>9317773012491</td>
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NetComm’s V210P VoIP Telephone Adaptor has been designed for residential and small business users to deliver predictable real-time voice quality over the Internet. It connects directly to any broadband modem and service (Cable or DSL) plus an account with a VoIP Service Provider.

Suitable for those who want the convenience of a landline (PSTN) combined with the savings of VoIP in one compact unit. The V210P delivers both VoIP and landline calls to a single handset, giving you the cost savings of VoIP with the security of a landline backup if and when you need it. The V210P also features QoS functionality to maintain call quality.

TECHNICAL SPECIFICATIONS

NETWORK PROTOCOL
- SIP v1 (RFC2543), v2(RFC3261)
- IP/TCP/UDP/RTP/RTCP
- IPv6/ICMP/ARP/RARP/SNTP
- TFTP Client/DHCP
- PPPoE Client
- HTTP Server
- DNS Client
- NAT/DHCP Server

TONE
- Ring Tone
- Ring Back Tone
- Dial Tone
- Busy Tone
- Programming Tone

CODEC
- G.711: 64k bit/s (PCM)
- G.723.1: 5.3k bit/s
- G.726: 32k bit/s (ADPCM)
- G.729A: 8k bit/s (CS-ACELP)
- G.729B: adds VAD & CNG to G.729

VOICE QUALITY
- VAD: Voice activity detection
- CNG: Comfortable noise generator
- LEC: Line echo canceller
- Packet Loss Compensation
- Adaptive Jitter Buffer

IP ASSIGNMENT
- Static IP
- DHCP
- PPPoE

CALL FUNCTION
- Call Hold
- Call Waiting
- Call Forward
- Caller ID
- 3-way conference

SECURITY
- HTTP 1.1 basic/digest authentication for Web setup
- MD5 for SIP authentication (RFC2069/RFC2617)

DTMF FUNCTION
- In-band DTMF
- Out-of-band DTMF
- SIP Info

NAT TRAVERSAL
- STUN
- SIP SERVER
- Registrar Server
- Outbound Proxy

CONFIGURATION
- Web Browser
- IVR/Keypad

FIRMWARE UPGRADE
- TFTP
- HTTP

AUTO PROVISIONING
- HTTP
- TFTP

MODEM & FAX MODES
- G.711 fax/modem pass-through with fax/modem detection
- T.38 support

INTERFACE
- 1 WAN port interface
- 1 LAN port interface
- 1 FXS port interface
- 1 FXO port interface