IP2061NK
NoKSU IP Phone System

Features

- An superior cost-saving phone system without KTS/PBX box
- Powerful Plug-and-play design offers easy installation
- Support multiple SIP Trunks with excellent interoperability with Softswitch, IP Centrex, IP-PBX and general SIP Server
- Intelligent technologies with P2P and SIP protocol convergence
- Quality voice and high performance data through the phones
- Auto-provisioning and Web management for ISP/ITSP carrier
- Rich supplementary call services and phone features
- Support multiple call/line appearance
- Support Extended Dial Module (optional)

Tecom IP2061NK NoKSU IP Phone can support simple low-cost but very powerful VoIP phone system for small offices. No need of any iPBX or IP-KTS support, up to 16 IP phones can work together with most supplementary call features. And all the IP phones can share the same SIP trunklines to connect to the ISP/ITSP VoIP service network to communicate with outside/PSTN world.

With sophisticated design, the proprietary Peer-to-Peer protocol is implemented to build the virtual IP-KTS environment and offer the switching mechanism with SIP protocol in the same phone. This approach reduces the outbound traffic and demand of more SIP accounts. Of course, as well it generates a higher communication security that all internal calls will go through P2P channels and not be exposed to the Internet public network.

With this sophisticated design and their versatile features, this IP phone system will increase much productivity and gains better investment return. Since it supports both Plug-and-play and Auto-provisioning, users can install the phone easily without the assistance of technician or system installer. For connecting to outside/PSTN, it can work with different IP-PBX, soft-switch, IP-Centrex (like SIP-B), or SIP server. In addition, and Extended Dial Module (EDM) is optional to extend its powerful programmable call/phone features or perform more IP-PBX supplementary features.

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Specifications

System

- Maximal up to 16 client for one NoKSU IP phone system
- Maximal up to 4 SIP trunk support

Standards

- Peer-to-Peer Proprietary protocol and Peer Discovery
- IETE SIP V2 (RFC3261) standards
- RTP, RTCP
- Static IP, DHCP
- DNS, TFTP, FTP, HTTP
- Telnet
- SNTP, STUN, SNMP V2 (optional)

Call Features

- Answering Position (VAA/Operator/All Ring/DID)
- Auto Answer
- Call Blocking
- Call Conferencing (3-way)
- Call Forwarding
- Call Hold & Hold Reminder
- Call Transfer/Recall
- Calling Line Identification (Name and Number)
- Distinctive Ringing
- Message Waiting Indictor
- Multiple Line Appearances
- Paging
- Redial, Call log, Phone book, Web, Speed Dial, and On-hook dialing
- VAA/VM

Voice Handling

- Codec G..711 a/μ law, G.729
- Supports VAD, CNG, AGC and Acoustic Echo Cancellation
- Jitter buffering and packet loss concealment
- Full-duplex Speakerphone
- Enhanced voice quality for handset, headset and hands-free

Phone Functions

- Call Record (Answered, missed, dialed)
- Call Timer & Duration
- Handsfree/headset Support
- Last Number Redial
- Mute
- Phone book
- Phone lock/unlock
- Speed Dial
- Volume Control

Voice Mail Sub-System

- Message Play, Stop, Next, Previous, Delete
- Message Waiting Indication
- Personal greeting Recording and Playback
- Greeting for Auto-Attendant
- Password change
- Audio storage in IP phone

Auto-Attendant Sub-System

- Call Answer Control
- Day/Night Service – Manual/Auto switch
- Dial by extension
- Escape to Attendant
- Greetings & prompt user-configurable

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Network & Security
- QoS: DSCP
- Admin/User Password
- Reset Admin Password
- P2P Signal Encryption

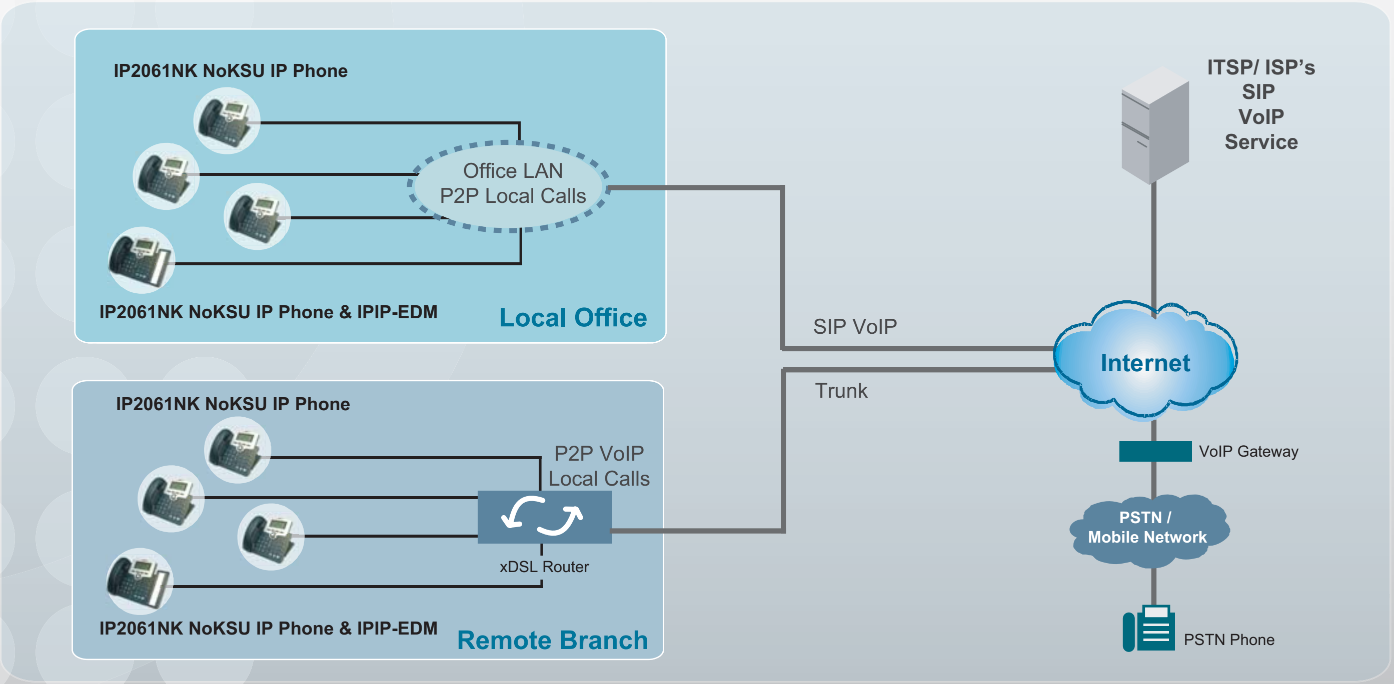
Configuration and Management
- SIP Trunk control/status on Line 1 ~ 4 keys
- Programmable call/phone features on EDM module
- Security and encryption support
- Remote and local configuration
- Web management
- Plug & Play for easy user installation and configuration
- Reset to factory default
- System clock setting automatically through SNTP server
- Engineering trace log (syslog)
- Software Upgrade through TFTP/HTTP serer
- Auto-provisioning
- Backup/Restore configuration to /from PC
- Reset Phonebook and Configuration
- Customer Differentiation/Protection support
- Phone Number Assignment

Display
- 128 x 64 graphic LCD display
- LED: MWI, Line 1 ~ 4, SPEAKER, MUTE, HEADSET
- Multi-language support (optional)

Keys
- Context-sensitive soft-keys & navigator keys
- Speaker, Mute, Headset Hold, Transfer, Conference, Message, Phonebook, Redial
- Line keys (1 ~ 4) for SIP Trunks
- Optional EDM module supported for more function keys (with LED)

I/O Ports
- Dual Base-T/100 Base-TX RJ-45 ports
- Headset port with RJ-9 connector
- EDM module connector

Power Supply
- Poe 802.3af built-in
- 5V DC/1A switching power adapter



TECOM
Converging Your Networks
TECOM CO., LTD.
23 R&D Road 2, Science-Based Industrial Park Hsin-Chu, Taiwan, R.O.C.
Tel : 886-3-5775141 Fax :886-3-5776854 web site: http://www.tecomproduct.com