SIP@Net – Call processing software

for SIP@Net server

At a Glance

- Uniform open software platform for SIP@Net server
- Full server based communication solution
- Open connectivity for SIP, IP, applications and networks
- Rich set of features designed for wired and non-wired users
- Tight functional integration with iS3000 hybrid communication solution

SIP@Net is a uniform software platform that supports the SIP@Net Server communication solution. The architecture of the SIP@Net provides IP based telephony.

SIP@Net provides customers with voice and networking features that are essential for conducting business in an effective and reliable way. It is a uniform software platform designed for the SIP@Net server as well as for iS3000 hybrid communication system enabling organisations of all types and sizes to combine SIP-based telephony with a broad range of traditional voice features.

Empowered by Innovation

Key features

- Uniform open software platform for the SIP@Net server communication system
- Rich set of features designed for wired and non-wired users
- Sophisticated features for group and manager/secretary arrangement
- Multiple features for operators
- Open connectivity for SIP, IP, applications and networks
- Software platform for NEC Unified Solutions applications
- Open interfaces for third party applications
- Tight functional integration with system management tools and maintenance features

Applications supported by SIP@Net are designed to meet specific user needs. They include:

- Announcement services
  - Voice Processing
  - Unified Messaging
  - Voice Mail
  - Interactive Voice Response (IVR)
- Automatic Call Distribution (ACD)
- Call logging (for example FDCR)
- Communication portals
- Computer Telephony Integration (ECMA CSTA and TAPI)
- Contact Centres
- Fixed Mobile Convergence
- IP/DECT/VoWLan cordless communication
- Messaging
- Microsoft: Office Communications Server and Exchange
- Presence management
- Private and multi-vendor networking (via IP networks and Gateways)
- Reachability
- System and network management

These applications are fully integrated within the SIP@Net software. They are each described in separate datasheets, while this datasheet provides an overview of SIP@Net functionality, when installed on a server. The features are listed according to different types of users.

Features

Generic extension features

All of the features mentioned below can be assigned to individuals or groups of extension numbers, allowing each extension to have its own authorisation profile. This applies to IP and SIP@Net Mobility Access (SMA) extensions.

- Abbreviated dialling:
  - individual
  - per user group
  - common pool
- Add-on conference (3-parties):
  - via CSTA interface
  - local on SIP terminal
- Automatic ring back:
  - on busy
  - on next use
  - cancel ARB
  - multiple ARB
  - ARB protected
  - ARB after diversion(s)
- Break-in:
  - party (ticker-tone)
  - break-in protection
- Call forwarding:
  - unconditional (follow-me)
  - on no answer
  - when absent
  - when busy
  - when not reachable
  - multi-hop call diversion
- Call waiting indication
- Camp-on busy:
  - automatic (destination profile)
  - call offer (originator profile)
  - music on camp-on busy
- Connected Party Display [1)]
- Cost Centre Dialling
- Distinctive ringing [2)] for:
  - internal calls
  - external calls
  - automatic ring back
  - emergency call
  - message waiting
- DDI barred
- Do not disturb:
  - routing to operator
  - user deactivation
  - deactivation after time-out
  - by-pass by operator

- Enquiry/call hold [2)]:
  - music on hold
- Follow-me:
  - activation from originating extension
  - activation from destination
  - originator dialling from destination
  - to fixed destination
  - follow-me protected
  - external follow-me (20 digits)
- General facility cancel
- Message waiting [3)]
- Personal Identification Dialling:
  - personal
  - network wide
- Secret Call:
  - calling line identification permanently restricted
  - connected line identification permanently restricted
- Selective diversions, distinguish on:
  - internal callers
  - external callers
- Shuttle/transfer [2)]:
  - call transfer before or after answer
  - shuttle between 2 parties
- Single-digit dialling
- Traffic class selection:
  - day/night traffic class
  - upgrade/downgrade traffic class by password dialling
  - password DDO (per call)
- Traffic class assignment:
  - per extension
  - per incoming gateway line
  - up to 8 traffic classes
- Twinning

1) For SIP@Net Mobility Access: operator dependent
2) Not for SIP@Net Mobility Access
3) Audible signal for SIP@Net Mobility Access

IP extension features

In addition to the generic extension features, the following features are available for the Polycom Soundpoint IP terminal range and Baseline Pro SIP:

- Automatic answering [4)]
- Callers-list (name/number logging on no answer)
- Connected number display
- Date and time display
- Desk sharing [4)]
- Directory
  - Local Directory Dialling
  - Central Directory via browser [4)]
- Display of own number
- Facility monitoring (LED icons):
  - automatic ring-back
  - do not disturb
  - follow-me
  - message waiting
- Distinctive ringing for:
  - internal calls
  - external calls
- Interfaces (Headphone) [4)]
- Message waiting indication
- Name dialling
- Programming of keys and terminal settings:
  - local by user
  - remote by means of download
- User-to-user text messaging [4)]

4) On Polycom sets only

SIP IP extension standards

SIP extension interfaces are based on the following SIP RFC standards:

- Basic Call handling (RFC 3261)
- Busy Lamp Appearance(draft-anil-sipping-bla-02)
- CLI and name display (RFC 3261)
- Connected Party Information (draft-ietf-SIP-identity)
- Diversion Header (draft-levy-SIP-diversion-08)
- DTMF support (RFC 2833)
- Fax (T38,G711)
- INFO handling (post-dialling) (RFC 2976)
- Instant messaging (RC 3428)
- Message waiting (RFC 3842)
- Payload formats in SDP (RFC 3555)
- Registration/Authentication (RFC 2617, 3216, 3261 section 22)
- RTP (RFC 3550)
- SDP handling (RFC 2327, RFC 3264)
- Secure Realtime Transport Protocol (SRTP) (RFC 3711)
- Security Description for Media Streams (RFC 4568)
- Session Guarding (RFC 4028)
- SIP Reliability (RFC 3262)
- Specific event notification (RFC 3265)
- Transfer (RFC 3515,RFC 3891)
- Transport Layer Security (TLS) (RFC 2246)

IP DECT cordless extension features

In addition to the generic extension features, the following features are available for cordless IP (DECT) extensions:

- Call duration display
- Call forwarding on no answer
- Call forwarding not reachable
- Calling number display
- Call waiting indication
- Central directory and status information
- Display of own number
- Distinctive ringing for:
  - internal calls
  - external calls
- Messaging
- Multi-site subscription
- Portable sharing
- Seamless handover
- Single number services
- Speech encryption
- Twinning

VoWLAN cordless extension features

In addition to the generic extension features, the following features are available for cordless IP (VOWLAN) extensions (depending on the specific IP business phone model):

- Call forwarding on no answer
- Calling number display
- Display of own number
- Messaging
- Multisite subscription
- Seamless handover
- Speech encryption
- Twinning

Group features

Groups may consist of IP cordless and IP extensions, freely mixed. Depending on the type of telephone, additional features include:

- Absent/present switching:
  - by extension user
- Absent-status indication (LED, display(icons))
- Announcements on empty group
- Call diversion on empty group
- Call pick-up:
  - individual extension
  - group call
  - unrestricted
- Camp-on busy queuing
- Chaining of group diversions
- Do not Disturb
- Follow-me
- Group follow-me
- Group hunting mechanism:
  - cyclic (round-robin)
  - linear (home hunting)
  - parallel (multiple ringing)
- Group park
- Group status display
- Monitoring absent/present status
- Monitoring idle/busy/ringing status
- Music-on-Hold
- Private Park

Manager/secretary features

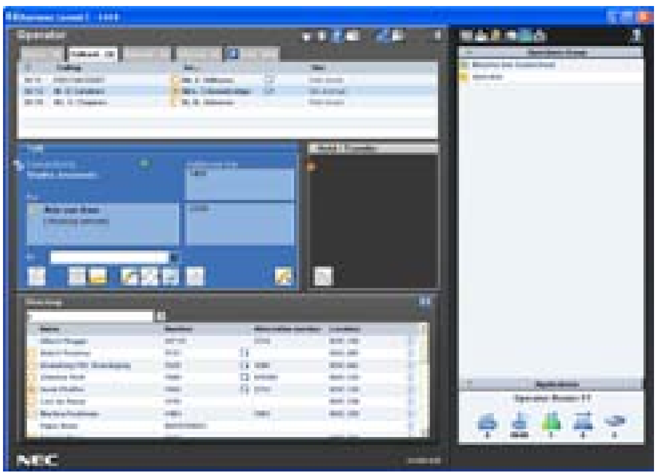
The following features are available – depending on the type of telephone – for users configured in a manager/secretary group:

- Absent overrule by secretary
- Absent/present switching by:
  - manager
  - secretary
- Alternative secretary:
  - break-in override
  - absent overrule
- Break-in override by secretary
- Break-in protection
- Call diversion to secretary on busy/absent
- Camp On Busy
- Intercom call
- Multi-call park (multi-hold)
- Multi manager/multi secretary arrangements
- Multi-line answering:
  - selective answering
  - internal/external call indication
- Private number
- Status monitoring:
  - busy/idle/ringing
  - absent/present

Operator features

Availability of features listed below are based on the Business ConneCT Operator application.

- Abbreviated dialling:
  - individual
  - user group
  - common pool
- Announcements on calls waiting for answer:
  - call to operator (per assistance group, operator or incoming route)
  - non-DDI calls
  - unsuccessful DDI call
- Break-in
- Call handling:
  - automatic return to operator
  - by-pass code (for call incompatibility)
  - transfer with announcement
  - release both parties
- Call identification:
  - calling number/name display
  - called number/name display
  - display last called party
- Call status display:
  - answered/busy/ringing/don't disturb/fail
- Directory Dialling
- Extension status monitoring
- Instant Messaging
- Password protection
- Queue features:
  - name display
  - long waiting call indication
  - selective call pick-up
- Queue types:
  - for incoming trunk calls
  - for recall and repeat calls
  - for internal calls
  - park queue



Night service

The night service can be assigned to any type of telephone set. Features include:

- Automatic absent switching:
  - on ringing time-out to night extension
- Automatic repeat:
  - on busy night extension
  - on non-answering night extension
- Busy override by night extension
- Call forwarding to higher night extension:
  - on absent
  - on busy
- Follow-me on night extensions
- Metering account for night service assisted calls
- Night extension groups
- Night extension hierarchy:
  - individual night extension
  - sub-common night extension (per incoming route)
  - main-common night extension (system-wide)
- Traffic-class upgrading and downgrading:
  - by code dialling
  - by traffic class switch
  - by system clock

Generic trunk features

The trunk features listed below are available for SIP interfaces:

- Alternative routing on congestion
- Barring external numbers
- Bundle splitting
- DDI-fail diversion on:
  - ringing time-out
  - number-unknown
  - busy extension
  - unsuccessful DDI call
- Default CLI
- Digit conversion
- Incoming DDI traffic:
  - digit conversion
  - DDI-fail to operator
- Least Cost Call Routing:
  - time-of-day
  - class of service
  - per user type (normal, priority or operator)
- Mobility Access
- Number analysis per trunk group
- Overflow on outgoing routes:
- Time-break, budget-break protected
- Toll-ticketing

IP SIP trunk standards

SIP@Net supports trunking with SIP-based Operators according to the following standards:

- Basic Call handling (RFC 3261)
- CLI and name display (RFC 3261)
- Diversion Header (draft-levy-SIP-diversion-08)
- DTMF support (RFC 2833)
- Fax (T38, G711)
- Payload formats in SDP (RFC 3555)
- QoS IEEE 802.1Q
- Registration/Authentication (RFC 2617, 3216, 3261 section 22)
- RTP (RFC 3550)
- SDP handling (RFC 2327, RFC 3264)
- Secure Realtime Transport Protocol (SRTP)(RFC 3711)
- Session Guarding (RFC 4028)
- SIP reliability (RFC 3262)
- Transport Layer Security (TLS) (RFC 2246)

Private Networking features

The SIP@Net offers networking capabilities for pure iS3000 server based networks, as well as networks with legacy iS3000 hardware platforms. The offered functionality is as follows:

Extension features

- Abbreviated dialling
- Automatic camp on busy (COB)
- Basic telephone call
- Break-in protection
- Call diversion:
  - follow-me active (immediate)
  - on no reply
  - on busy
  - to external (immediate)
- Call hold (start enquiry)
- Calling-line/name display
- Call offer (COB by originator)
- Call waiting (COB by destination)
- Connect-line/name display
- Intercom
- Message-waiting indicator
- Number-presentation restriction
- Three-party conference
- Transfer before answer (new party)
- Twinning
- User-to-user text messaging

System features

- Distinctive ringing
- Free numbering
- Loop avoidance
- Night-assistance routing
- Route optimisation
- Trunk identity
- Uniform numbering plan (12 digits):
  - closed
  - open

SIP@Net Server applications

The SIP@Net offers application support for:

- Business ConneCT
  - Employee
  - Operator
  - Agent
- Management
  - MA4000
  - MAS 9
- iSMobile
- Messenger@Net
- Unified Messaging

Computer Telephony Integration (CTI)

For CTI applications the following CTI features can be activated:

- Call answer
- Call diversion
- Diversion, Fallback reason
- Group manipulating
- Group monitoring
- Hold call
- Maintenance event reporting
- Message waiting
- Twinning

Maintenance features

OM (Operational Maintenance) procedures are provided via PC based tooling. Enhanced features can be obtained through system management modules for System and Network management.

- Alarm handling
- Fault diagnostics
- File Transfer Protocol Support (FTP)
- On-line journaling
- Operational Maintenance
  - moves and changes
  - facilities
  - traffic classes
  - user profiles
- Password protection
- Performance analyses
- SNMP
- Software download
- System assurance reports
- System identification
  - local/remote read out of software identification
- Test Call
- Traffic measurement

Expert Services

SIP@Net is fully supported by our Expert Services. This extensive portfolio of services provides the insight and support needed to get the most out of equipment and applications. The services offered comprise advice, design, customisation, integration, training, maintenance, continuous optimisation and Business Partner services.

Technical data

Programming language

SIP@Net is programmed in the object-oriented programming language C++.

Operating system

SIP@Net server runs on Windows server operating system.

Environment

The features listed for SIP server are available when installed on a server platform. Connection to legacy infrastructure and devices requires suitable gateways.

Field upgradeable

New versions can be downloaded on site as well as remote.

UNIVERGE®360 is NEC's approach to unifying business communications. It places people at the center of communications and delivers on an organization's needs by uniting infrastructure, communications and business.

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