



SIP@Net

SOPHO iS3000 – SIP@Net Call processing software

Single and fault tolerant processors

At a Glance

- Uniform open software platform for SOPHO iS3000 communication system
- Open interfaces for third party applications
- High feature level on Polycom SIP terminals
- Open connectivity for SIP, IP, applications and networks
- Multiple features for operators

SIP@Net is a uniform software platform that supports the SOPHO iS3000 communication system. The architecture of the SOPHO iS3000 is designed to provide both IP telephony and traditional PBX solutions, resulting in a fully hybrid system that combines the best of IP packet switching and traditional TDM technology.

SIP@Net is a key element in providing a broad range of critical voice features and functions within a converged network environment. SIP@Net introduces SIP call processing software and forms the basis of the iS3000 platform's migration to IP converged networks.

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 SOPHO iS3000 hybrid communication system and the iS3000 SIP extension server, enabling organisations of all types and sizes to combine SIP-based telephony with a broad range of traditional voice features.

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Key features

- Uniform open software platform for the SOPHO iS3000 communication system
- Open connectivity for SIP, IP, applications and networks
- Rich set of features designed for wired and non-wired users
- High feature level on Polycom SIP terminals
- Sophisticated features for group and manager/secretary arrangement
- Multiple features for operators
- Software platform for NEC Philips applications
- Open interfaces for third party applications
- Tight functional integration with system management tools and maintenance features

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

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



These applications are fully integrated within the SOPHO SIP@Net software. They are each described in separate datasheets, while this datasheet provides an overview of SOPHO SIP@Net functionality, when installed on the iS3000 platform. 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 ISDN, IP and SOPHO Mobility Access (SMA) extensions.

- Abbreviated dialling:
 - individual
 - per user group
 - common pool
- Add-on conference, 3-parties: ¹⁾
 - internal or external parties
 - in-band ticker tone (optional)
- Automatic ring back:
 - on busy
 - on next use
 - cancel ARB
 - multiple ARB
- ARB protected
- ARB after diversion(s)
- Automatic trunk location:
 - by automatic ring back
 - by camp-on busy
- 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 ¹⁾
- Calling Number Display ²⁾
- Camp-on busy:
 - automatic (destination profile)
 - call offer (originator profile)
 - music on camp-on busy
 - indication of position in queue
- Cost Centre Dialling

- Creation of DNR from set (for installation) ¹⁾
- DDI barred
- Desk sharing
- Dialed intercom
- Distinctive ringing ³⁾ for:
 - internal calls
 - external calls
 - automatic ring back
 - emergency call
 - message waiting
- Do not disturb:
 - routing to operator
 - user deactivation
 - deactivation after time-out
 - by-pass by operator
- Enquiry/call hold/transfer/shuttle
 - progressive enquiry
- Follow-me:
 - activation from originating extension
 - activation from destination
 - originator dialling from destination
 - to fixed destination
 - follow-me protected
 - external follow-me (20 digits)
 - to paging
- General facility cancel
- Hot-line: ¹⁾
 - immediate
 - delayed
- Last External Number Repeat (LENR)
- Malicious Call Trace ¹⁾
- Message waiting
- Multi-party conference (up to 8 parties)
- Multi-party listen in (up to 16 users)
- Personal Identification Dialling
 - personal
 - network wide
- Post-dialling
- Secret Call:
 - dial-up secret call
 - calling line identification permanently restricted
 - connected line identification permanently restricted
- Selective diversions, distinguish on:
 - internal callers
 - external callers

- Shuttle/transfer:
 - call transfer before or after answer
 - shuttle between 2 parties
 - shuttle with forced release
- 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 trunk line
 - up to 8 traffic classes
- Twinning
- Voice logging



Digital extension features

In addition to the generic extension features, the following features are available (depending on the specific digital business phone model):

- Automatic answering
- Call charge display:
 - in currency/pulses
 - during/after the call
- Callers-list (name/number logging on no answer)
- Calling name/number display
- Camp-on busy queue position display
- Connected number display
- Date and time display:
 - real-time system clock
 - ISDN synchronisation
- Desksharing
- Directory
 - Local Directory Dialling
 - Central (in switch) Directory Dialling for internal and external names
- Display diversion reason of received call
- Display of own number

¹⁾ not for SIP extension interfaces

²⁾ for SOPHO Mobility Access extensions depending on telecom provider

³⁾ not for SOPHO Mobility Access extension

- Facility monitoring (LED):
 - automatic ring-back
 - do not disturb
 - follow-me
 - call-waiting
 - camp-on busy call
- Intercom call:
 - both-way
 - microphone mute
- Interfaces (V.24, audio, aux)
- Message waiting indication
- Name dialling
- Off-line number preparation
- Programming of keys and terminal settings:
 - local by user
 - remote by means of download
- Route indication
- User-to-user text messaging:
 - send message during a call
 - answer-back message

SIP IP extension features

In addition to the generic features, the following features are available on the Polycom terminal range:

- Automatic answering
- Callers-list (name/number logging on no answer)
- Calling name/number display
- Connected number display
- Date and time display:
- Desksharing
- Directory
 - Local Directory Dialling
- Display of own number
- Facility monitoring (LED):
 - automatic ring-back
 - do not disturb
 - follow-me
 - Message waiting



- Intercom call:
 - both-way
- Interfaces (audio, aux)
- Message waiting indication
- Name dialling
- Off-line number preparation
- Programming of keys and terminal settings:
 - local by user
 - remote by means of download
- User-to-user text messaging

The following standards are supported on the SIP IP extension interfaces:

- 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 (RFC 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)
- SIP Reliability (RFC 3262)
- Specific Event Notification (RFC3265)
- Transfer (RFC 3515,RFC 3891)
- Transport Layer Security (TLS) (RFC 2246)

Cordless extension features

In addition to the generic extension features, the following features are available for cordless IP and DECT extensions, depending on the type of telephone set:

- Call duration display
- Call forwarding on no answer
- Call forwarding not reachable
- Calling number display
- Central Directory
- Date and Time display
- Display of own number
- Hot-line mode
- Messaging
- Multi-site handover ¹⁾

1) not for SIP extension interfaces

- Multi-site roaming
- Multi-site subscription
- Portable sharing
- Seamless handover
- Single number services
- Site detection
- Speech encryption
- Twinning



Group features

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

- Absent/present switching:
 - by extension user
 - by group supervisor
- Absent-status indication by:
 - special dial-tone ¹⁾
 - LED, display text
- Announcements on empty group
- Call diversion on empty group
- Call pick-up:
 - individual extension
 - group call
 - unrestricted
- Camp-on busy queuing
- Camp-on busy queue length display
- 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 (9 positions)

- Group status display
- Monitoring absent/present status
- Monitoring idle/busy/ringing status
- Multi-line answering:
 - selective answering
 - internal/external call indication
- Music-on-Hold
- Private Park (2 positions)

1) not for SIP extension interfaces

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
- Multi-call park (multi-hold)
- Multi manager/multi secretary arrangements
- Private number
- Status monitoring:
 - busy/idle/ringing
 - absent/present
- Single-key intercom call ¹⁾

1) not for SIP extension interfaces

ISDN extension features

These features are available for third-party ISDN terminals.

The applicable abbreviations and ETSI standards are indicated in brackets:

- Advice of call charge (AOC, ETS 300181):
 - during the call (AOC-D, ETS 300 179)
 - after the call (AOC-E, ETS 300 180)
- Basic Call (ETS 300 192)
- B-channel bundling
- Call Hold (CH, ETS 300 139)
- Calling Line Identification Presentation (CLIP, ETS 300 191)
- Calling Line Identification Restriction (CLIR, ETS 300 191)
- Connected Line Identification Presentation (COLP, ETS 300 191)
- Connected Line Identification Restriction (COLR, ETS 300 191)
- Call Transfer (CT, ETS 300 260))
- Call Waiting (CW) ETS 300 056

- In-Call Modification (ICM)
- Keypad protocol for facility access (ETS 300 190)
- Multiple Subscriber Number (MSN, ETS 300 052)
- Sub-addressing (SUB, ETS 300 059))
- Terminal Portability (TP, ETS 300 055)

Operator features

Availability of features listed below depends on specific SuperVisor operator console models:

- Abbreviated dialling:
 - individual
 - user group
 - common pool
- Alarm signalling: ⁴⁾
- Announcements on calls waiting for answer:
 - call to operator (per assistance group, operator or incoming route)
 - non-DDI calls
 - unsuccessful DDI call
- Auto-attendant (see Voice Processing Facilities)
- Automatic Ring Back
- Break-in: on busy extension
- Extension/status monitoring
- Call handling:
 - automatic return to operator
 - call splitting
 - by-pass code (for call incompatibility)
 - enquiry on trunks
 - give a line ⁴⁾
 - listen-in
 - outgoing call assistance
 - transfer with announcement
 - fast call transfer
 - release both parties



- Call identification:
 - calling number/name display
 - called number/name display
- Call status display:
 - answered/ busy/ringing/don't disturb/fail
 - DDO-line reservation for operators
- Directory Dialling
- Flexible operator:
 - de-centralised operator assistance
 - centralised operator assistance
 - distributed operator assistance
- Messaging to/from operator:
 - manual messaging by operator
 - automatic messaging by user
 - user-to-operator messaging
- Multi-party conference:
 - operator supervision
 - up to 8 parties
- Multi-tenant operator
- Night service:
 - automatic switch-over to night service
 - manual switch-over to night service
 - night mode display ⁴⁾
- Password protection of login
- Queue features:
 - real-time queue thermometers
 - long waiting call indication
 - priority call indication
 - urgent call monitoring ⁴⁾
 - overflow extension on overloaded queue
 - Selective call pickup ⁵⁾
- Queue types:
 - for incoming trunk calls with/with out priority
 - for recall and repeat calls
 - for internal calls with/without priority
 - for calls on hold

⁴⁾ not on Business ConneCT Operator

⁵⁾ for Business ConneCT Operator only

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
- Call urging tone during queuing on a busy night-extension
- Follow-me on night extensions
- Metering account for night service assisted calls
- Night extension groups
- Night extension hierarchy:
 - individual night extension (non-DDI trunks)
 - 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 various signalling types and access interfaces:

- DDI-fail diversion on:
 - ringing time-out
 - number-unknown
 - busy extension
 - dialling time out
 - unsuccessful DDI call
- Default CLI (on 'in dialling barred')
- Incoming DDI traffic:
 - digit conversion
 - DDI-fail to operator
- Incoming non DDI traffic:
 - permanent line extensions
 - operator assistance
- Malicious Call Trace
- Mobility Access
- Overflow on outgoing routes:
 - Direct Dialling Out (DDO)
 - tax-metering
 - tax-metering of assisted calls
 - post dialling
 - keytone to dial pulse conversion and vice versa
 - busy
 - congested route
- Time-break, budget-break protected
- Toll-ticketing

IP trunk features

SOPHO 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)
- SIP reliability (RFC 3262)
- Transport Layer Security (TLS) (RFC 2246)

ISDN trunk features

The ISDN trunk features are available for different access protocols, i.e. 1TR6 (Germany), DASS2 (UK) and Euro-ISDN:

- Advice on call charge (AOC, ETS 300181):
 - during the call (AOC-D, ETS 300 179)
 - after the call (AOC-E, ETS 300 180)
- Call Completion on Busy Subscriber (CCBS, ETS 300 359)
- Calling Line Identification Presentation (CLIP, ETS 300 089)
- Calling Line Identification Restriction (CLIR, ETS 300 190)
- Calling Name Identification Presentation (CNIP, ETS 300 238)
- Connected Line Identification Presentation (COLP, ETS 300 092)
- Connected Line Identification Restriction (COLR, ETS 300 193)
- Direct Dialling In (DDI, ETS 300 064)
- Direct Dialling Out (DDO, ETS 300 059)
- In-Call Modification (ICM)
- ISDN Addressing (ADDR, ETS 300 189)
- Malicious Call Identification (MCI, ETS 300 130)
- Sub-addressing (SUB, ETS 300 059)
- Sub-addressing (SUB, ETS 300 192)
- Terminal Portability (TP, ETS 300 055)

Trunk routing features

The trunk routing features are available for various signalling types (ISDN, MFC, E&M, LD, IP etc.) and access interfaces:

- Alternative routing on congestion
- Barring external numbers
- Bundle splitting
- Delayed outgoing seizure
- Digit conversion
- Enquiry on trunks
- Least Cost Call Routing:
 - time-of-day
 - class of service
 - per user type (normal, priority or operator)
- Mercury (in)direct access (UK)
- Number analysis per trunk group
- Privileged route selection

Computer Telephony Integration (CTI)

Together with a CTI application, running on a PC, connected to the system's Ethernet connection, a number of 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 OM terminals and operator consoles. Enhanced features can be obtained through system management modules for System and Network management.V.24 and IP interfaces are available.

- Alarm handling
- Built-in protocol trace (ISDN, QSIG, DPNSS)
- Fault diagnostics
- File Transfer Protocol Support (FTP)
- On-line journaling
- Operational Maintenance
 - moves and changes
 - facilities
 - traffic classes
 - user profiles



- Password protection
- Performance analyses
- Software download
- System assurance reports
- System identification
 - local/remote read out of hardware & software identification of boards
- 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

Within the iS3000 Hybrid Communication System, SIP@Net runs on the embedded operating system pSOS.

Environment

The features listed are available for the iS30X0 series, iS3000 Single and Fault Tolerant processor models.

Field upgradeable

New versions can be downloaded into the system during operation. SIP@Net supersedes Call@Net call processing software.

About NEC Philips Unified Solutions: NEC Philips Unified Solutions is a leading provider of comprehensive communications solutions to organisations of all types and sizes. NEC Philips is part of NEC's global enterprise communications activity, ranked nr 1 in worldwide enterprise communications. Its advanced IP-based solutions unify products, applications and services, enabling organisations to communicate and collaborate more effectively, efficiently and reliably. As Microsoft Gold Partner, NEC Philips specializes in providing real-time Unified Communications (UC) and collaboration solutions. NEC Philips Unified Solutions is headquartered in Hilversum, the Netherlands, and serves its customers through a global network of sales organisations and business partners. For more information: www.nec-philips.com.

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