SAMoffice8

SAMIP® Communication Server

SAMoffice 8 is the integrated solution of the SAMIP® family developed by SELTA to meet the communication needs of medium and large companies. The system allows widespread use of all the most advanced telephone functions available on the market today and is designed to guarantee a high-performance communication service for any type of telephone user, both IP and traditional. The flexibility and modularity of the elements make SAMoffice 8 suitable for companies that want to integrate the data network with the voice network and that pay great attention to cost optimization and the improvement of internal processes. **SAMoffice 8** is a communication server that offers each user the same service independently of the device and applications used by the user. The integration of SAMoffice 8 with vertical applications completes the system by linking it to the activities, processes and specific needs of the company and the market.

MAIN FEATURES

SAMoffice 8 has been designed to connect to any type of traditional telephone line (ISDN or PSTN) as well as to VoIP trunk lines provided by ISPs and SIP carriers. The modularity of the system allows you to configure **SAMoffice 8** according to the business needs of the moment while ensuring the ability to respond to the future operational needs of the organization. VoIP resources (up to 192 simultaneous communication channels) can be used for networking in multi-site scenarios and to use all the services available on SELTA IP terminals. The integrated SIP Proxy makes it possible to connect SIP standard telephones and clients installed on PCs, smartphones or tablets, which can also be reached via the Wi-Fi network. The large number of features offered by **SAMoffice 8**, such as Voice Mail, Automated Attendant, Conference Server service, Attendant console with accessibility options for blind or low-vision users, TAPI-based services, etc. allow you to meet all the communication needs of a modern organization. SAMIP® systems can perform advanced gateway functions in combination with the SELTA BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

MAIN FUNCTIONS

VoIP
Allows immediate use of dedicated resources for networking and IP telephony, or for cost containment thanks to the connection with SIP Carriers.

Modular
Grows flexibly based on actual business needs.

Complete
It has a wide range of telephone services for any user, fixed or mobile.

Multi-sites
Allows geographic distribution of corporate communications via IP or TDM QSIG links with branches, agents, mobile personnel, etc.

Open
The system seamlessly integrates with business applications and IT infrastructure to maximize the return on your technology investment.

Simplified management
Administrators can manage the system via web or dedicated client.

SYSTEM CAPABILITY

- External lines
  - up to 32 BRI T0/S0 ISDN Ports
  - up to 64 FX0 Ports (analog trunk lines)
  - up to 496 FXS Ports (analog extensions)
  - up to 4 PRI E1/T1 ISDN Ports
- Telephone extensions
  - up to 384 SELTA IP 51xx Series extensions
  - up to 384 IP Netfon Bluelight extensions
  - up to 384 IP SIP extensions
  - up to 496 analog extensions
- DSP resources
  - up to 192 VoIP channels with DSP Farm units

TECHNICAL SPECIFICATIONS

- Dimensions and weight
  - subrack (mm): H.135 x W.490 x D.410; 7 Kg
- Mounting
  - 19" data rack (3U for each subrack)
- Connectors
  - connection to the LAN network: 10/100 BASE-T Ethernet port switch, IEEE 802.3 / 802.3u, AutoSense, Automatic cable crossover (MDI / DI-X), PoE 802.3af
  - AUX port for inputs and outputs to the field (2 DIN, relay output)
  - COM-RS232 port
  - 2 LVDS interfaces for Extended (expandable up to 8)
- Power Supply
  - integrated AC / DC power supply
  - AC IN: 90-264 Vac 50/60 Hz 2.4 / 0.8A
  - integrated batteries (optional)
  - DC IN (opt.): Input of power supply voltage -48Vdc (-38Vdc ÷ -60Vdc), max current for single subrack 4.5A
- Consumption
- Estimated data for the base system:
  - SAMoffice8 Full IP configuration: 40 W
  - SAMoffice8 TDM configuration: 86 W
- Peak traffic consumption: 165 W

SYSTEM DATA

- BHCC
  - up to 30.000 Busy Hour Call Completion
- Central unit
  - Intel Xscale main processor, matrix
  - 4096 channel TDM switching and 32 PCM wires, ARM auxiliary processor
- Operating system
  - Linux

CERTIFICATIONS

- CE marking (European Community directive 1999/05 / EC: ETSI EN 300 386; EN 55022; EN 55024; EN 61000-3-2; EN 61000-3-3; EN 60950)
- Interoperability on TIM's Hyperway network (network of SAMIP exchanges)
- PSTN interface (ETSI TBR-21)
- ISDN-PRA public network interface (ETSI TBR-4; ETS 300102; ETS 300125)
- ISDN-BRA Public Network Interface (ETSI TBR-3; ETS 300102; ETS 300125)
- Caller ID (ETS 300 778-1)
- Environmental tests (ETS EN 300 019 class 3.2 in use) performed in the internal laboratory

APPLICATIONS AND CTI
- TAPI 1st party (Microsoft TAPI2.2)
- TAPI 3rd party (Microsoft TAPI2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - Multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - Click to talk (VoIP)
  - Call me back
  - Co-browsing
- CRM-oriented
- Metrics reports
  - Call Flow Editor:
  - Integrated VM/AA
  - Facility UM
- Unified Messaging:
  - Voice2Email
  - FAX2Email
  - SMS2Email
- Integrated Hotel Services:
  - On the phone and P.O.
  - Web Hotel in browser
  - External PMS integration

VoIP FEATURES
- Signalling
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice encodings
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Jitter Buffer configurable
- QoS via DiffServ (RFC 2474/2597/2598)
- QoS via TOS (RFC791)
- H.323 gatekeeper support (ITU-T)
- Trunks lines backup
- Music on hold

VoIP AND ToIP SERVICES
- SIP gateway
  - analog / digital terminals
  - analog / ISDN lines (BRI / PRI)
  - QSIG network
  - S0 data
- H.323 gateway (ITU-T H.225.0)
- SIP network with protocol (RFC 3204)
- Fax T38 (ITU-T)
- Standard SIP terminal support
  - hardphone
  - softphone
- Support standard SIP Wi-Fi terminals (IEEE 802.11)
- Support hybrid SIP terminals Wi-Fi - 2/3 g (IEEE 802.11 / ITU-T IMT-2000)
- Standard H.323 terminal support
- Proprietary IP terminals for KTS services:
  - hardphone (NETfon)
- Internet connection to the company LAN with a NETfon phone:
  - telework
  - peripheral site
- Power-over- Ethernet (PoE) (IEEE 802.3af)

* Maximum equipment not reachable at the same time

	BASE	MAX
IP USERS	16	384
TDM USERS	16	496
AVAILABLE SLOTS	8	31
EXPANSION MODULES	-	3

LEGACY PBX SERVICES
- User termination management:
  - analog (CEI 103, TBR 21)
  - proprietary digital (SAEfon CL family) (CEI 103)
  - ISDN on S0 interface (ETSI TBR-3)
- Operator console (CEI 103):
  - supervisor Console
  - desktop Attendant console (Microsoft TAPI 2.2)
  - desktop Attendant console with Accessibility options for blind or low-vision users
- Management of urban line terminations:
  - analog (ETSI TBR-21)
  - ISDN PRA (ETSI TBR-4, 300 102, 300 125)
  - ISDN (BRA) (ETSI TBR-3, 300 102, 300 125)
- Management of private network terminations:
  - analog E&M
  - digital QSig (2Mbit/s) (ETS 300 011/ 170/ 171/ 172/ 173/ 237/ 238/ 260/ 261)
  - Legacy Class 5 Caller ID telephone services (ETS 300 778-1)
- ISDN supplementary services:
  - malicious call identification (ETS 300 128 / 129/130)
  - calling / Connected Line Identification (ETS 300 089/ 091/ 092 / ETS 300 094/ 096/ 097)
  - calling / Connected Line Identification Restriction (ETS 300 090/ 091/ 093/ 095/ 096/ 098)
  - hold and retrive (ETS 300196-1)
  - three-party conference (ETS 300 188-1)
  - taxation according to the “SIP pilot” and ETSI-DSS1 standards
  - remote reading of costs
- QSIG Services:
  - basic call (ISO / IEC 11572)
  - name identification (ISO IEC 13868)
  - diversion (ISO / IEC 13673)
  - call transfer (ISO / IEC13869)
  - call offer (ISO / IEC 14843)
  - call completion on busy subscriber (ISO/IEC13870)
  - call completion on no reply (ISO / IEC 13870)
  - path replacement (ISO / IEC 13874)
  - advice of charge (ISO / IEC DIS15050)
  - message waiting indication

TELEPHONE SERVICES FOR SIP USERS
- Calling Line Identification Presentation/Restriction (CLIP/CLIR)
- Connected Line Identification Presentation/Restriction (COLP / COLR)
- Call holding and retrieval (RFC 3261)
- Call forwarding (RFC 3261)
- Call transfer (RFC 3515/3891)
- Conference (DRAFT 00)
- Instant messaging (RFC 3428 partial)
- Video call (RFC2327)
- Multilinearity
- DTMF transmission / reception (RFC3261 / 2833)

UTILITY FOR SIP USERS
- Registration controlled by management system
- Hierarchical management of SIP users through service classes
- Controlled access to system resources:
  - outband output
  - abbreviated numbering
  - prohibited numbers
- Easy access to "trusted-host" SIP remote users
- Voice mail
- Group management in "SIP forking" mode (RFC 3261)
- Call forwarding
- Call transfer

MANAGEMENT AND PROVISIONING
- IP Management:
  - online and offline configuration
  - FW loading of units
  - control panels and peripherals via the web
  - uploading welcome messages via web
  - configuration release management facility
  - online and offline maintenance
  - online alarm collection
  - accounting and CDR
- Web Personal Provisioning
- Multi-site Management:
  - alarm
  - accounting
- SNMP-TRAP Service Support (RFC 1157)

SAMIP® systems can perform advanced gateway functions in combination with the SELTA BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications

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SELTA
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SAM4000E

SAMIP® Communication Server

SAM4000E is the integrated communication solution of the SAMIP® family for the Enterprise world and Critical Infrastructure market, designed to offer high reliability in the most critical conditions. **SAM4000E** is designed to provide high-performance communication services to a huge number of both IP and TDM users, managing up to a maximum of 2560 “line / trunk” ports. The hot redundancy of the main parts makes the **SAM4000E** apposite for “mission critical” applications, particularly in environments sensitive to the high availability requirements of the communication system. **SAM4000E** is the communication server for large companies, in which business processes cannot ignore the integration between voice and data, which guarantees each user the same service independently of the device and applications used. Thanks to its flexibility, **SAM4000E** adapts perfectly to multi-site or campus-type scenarios in combination with the other systems of the SAMIP® family. SAM4000E can be used in hosted/managed mode also thanks to the centralized management system designed to easily and transparently connect a large number of users spread over different locations. The integration of **SAM4000E** with vertical applications completes the system by tying it to the activities, processes and specific needs of the company and the market.

MAIN FEATURES

SAM4000E is designed to connect to any type of traditional telephone line (ISDN or PSTN) as well as to VoIP trunk lines provided by ISPs and SIP carriers. The system modularity permits you to configure the **SAM4000E** according to the business needs of the moment while ensuring the ability to respond to the future operational needs of the organization.
VoIP resources (up to 320 simultaneous communication channels) can be used for networking in multisite scenarios and to use all the services available on SELTA IP terminals. The integrated SIP Proxy makes it possible to connect SIP standard telephones and clients installed on PCs, smartphones or tablets, which can also be reached via the Wi-Fi network. The large number of features offered by SAM4000E, such as Voice mail, Automated Attendant, Conference Server service, Attendant console with accessibility options for blind or low-vision users, TAPI-based services, etc. allow you to meet all the communication needs of a modern organization.
SAMIP® systems can perform advanced gateway functions in combination with the SELTA BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

MAIN FUNCTIONS

VoIP
Allows immediate use of dedicated resources for networking and IP telephony, or for cost containment thanks to the connection with SIP Carriers.
Modular
Grows flexibly and scalable according to actual business needs.
Reliable
The system architecture includes the redundancy of the central unit, power supply and connections with the subracks to provide a high level of resiliency in critical situations.
Complete
It has a wide range of telephone services for any user, fixed or mobile.
Multi-sites
Allows geographic distribution of corporate communications via IP or TDM QSIG links with branches, agents, mobile personnel, etc.
Open
The system seamlessly integrates with business applications and IT infrastructure to maximize the return on your technology investment.
Simplified management
Administrators can manage the system via web or through a dedicated client.

SYSTEM CAPABILITY

External lines
- up to 60 BRI T0/S0 ISDN Ports
- up to 120 FX0 Ports (analog trunk lines)
- up to 2016 FXS Ports (analog extensions)
- up to 8 PRI E1/T1 ISDN Ports

Telephone extensions
- up to 512 SELTA IP 51xx Series extensions
- up to 512 IP NETfon Bluelight extensions
- up to 512 IP SIP extensions
- up to 2016 analogue extensions

DSP resources
- up to 320 VoIP channels with DSP Farm units

TECHNICAL SPECIFICATIONS

Dimensions and weight
Subrack (mm) : H.225 x L.490 x P.240; 9Kg
Mounting
on 19 " data rack (5U for each subrack)
Connectors
- Connection with the LAN network: 10/100 Base-T Ethernet port switch, IEEE 802.3 / 802.3u, AutoSense, Automatic Cable crossover (MDI / DI-X), PoE 802.3af
- AUX port for inputs and outputs to the range (2 DIN, relay output)
- COM-RS232 port
- 2 LVDS interfaces for Extended modules (expandable up to 8 and redundable)

Power Supply
- Redundant DC/DC converter
- DC IN: Input of the supply voltage -48Vdc (-38Vdc: -60Vdc) max current for single subrack 9A
- Integrated AC/DC power supply (opt.)

Consumption
- Estimated data for the base system:
- SAM4000E Full IP configuration: 97 W
- SAM4000E TDM configuration: 154 W
- Peak traffic consumption: 220 W

SYSTEM DATA

BHCC
Up to 30.000 Busy Hour Call Completion
Central Unit (redundant)
Intel Xscale main processor, 4096-channel, 32-wire PCM TDM switching matrix, ARM auxiliary processor
Operating system
Linux

CERTIFICATIONS

- CE marking (European Community directive 1999/05 / EC: ETSI EN 300 386; EN 55022; EN 55024; EN 61000-3-2; EN 61000-3-3; EN 60950)
- Interoperability on TIM's Hyperway network (network of SAMIP exchanges)
- PSTN interface (ETSI TBR-21)
- ISDN-PRA public network interface (ETSI TBR-4; ETS 300102; ETS 300125)
- ISDN-BRA Public Network Interface (ETSI TBR-3; ETS 300102; ETS 300125)
- Caller ID (ETS 300 778-1)
- Environmental tests (ETS EN 300 019 class 3.2 in use) performed in the internal laboratory

APPLICATIONS AND CTI

- TAPI 1st party (Microsoft TAPI 2.2)
- TAPI 3rd party (Microsoft TAPI 2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - Multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - Click to talk (VoIP)
  - Call me back
  - Co-browsing
- CRM-oriented
- Metrics reports
  - Call Flow Editor:
  - Integrated VM/AA
  - Facility UM
- Unified Messaging:
  - Voice2Email
  - FAX2Email
  - SMS2Email
- Integrated Hotel Services:
  - On the phone and P.O.
  - Web Hotel in browser
  - External PMS integration

VoIP FEATURES

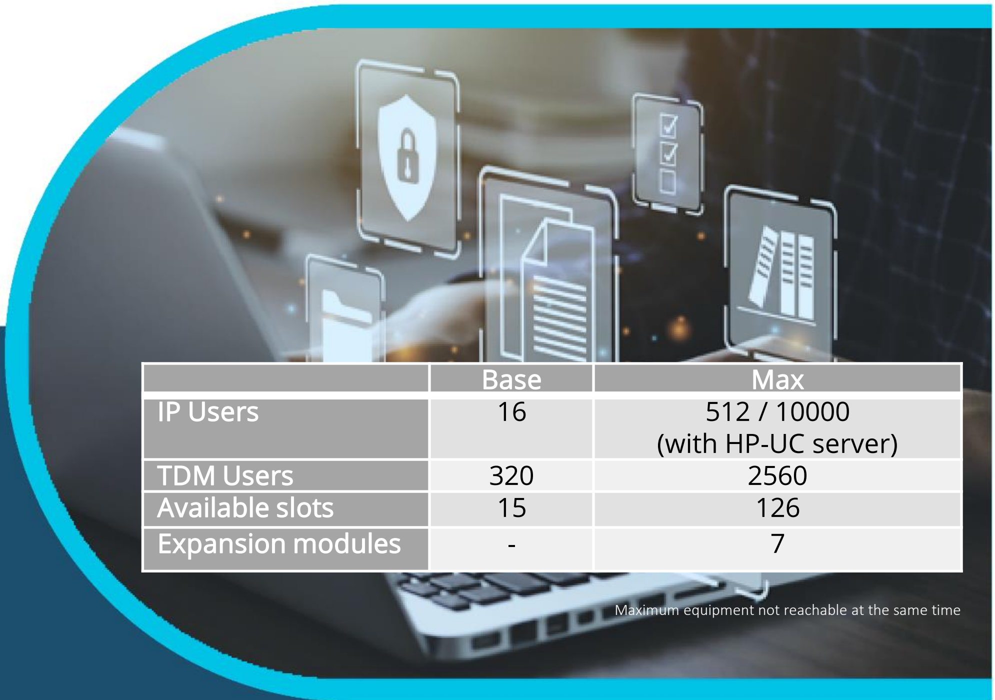
- Signalling
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice encodings
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Configurable Jitter Buffer
- QoS via DiffServ (RFC 2474/2597/2598)
- QoS via TOS (RFC791)
- H.323 gatekeeper support (ITU-T)
- Trunk lines backup
- Music on hold

VoIP AND ToIP SERVICES

- SIP gateway
  - analog / digital terminals
  - analog / ISDN lines (BRI / PRI)
  - QSIG networking
  - S0 Data
- H.323 gateway (ITU-T H.225.0)
- SIP networking with SIP-T protocol (RFC3204)
- Fax T38 (ITU-T)
- Standard SIP terminal support:
  - hardphone
  - softphone
- Wi-Fi terminal support (IEEE 802.11)
- Support hybrid SIP terminals

Wi-Fi - 2/3 g (IEEE 802.11/ITU-T IMT-2000)

- Standard H.323 terminal support (called base) (ITU-T)
- Proprietary IP terminals for KTS services:
  - hardphone (NETfon)
- Internet connection to the company LAN with a NETfon telephone:
  - telework
  - peripheral site
- Power-over-Ethernet (PoE)(IEEE 802.3af)



	Base	Max
IP Users	16	512 / 10000 (with HP-UC server)
TDM Users	320	2560
Available slots	15	126
Expansion modules	-	7

Maximum equipment not reachable at the same time

LEGACY PBX SERVICES

- **User termination management:**
  - analogue (CEI 103, TBR 21)
  - proprietary digital (SAEfon CL family) (CEI 103)
  - ISDN on S0 interface (ETSI TBR-3)
- **Operator console (CEI 103):**
  - supervisor Console
  - desktop Attendant console (Microsoft TAPI 2.2)
  - desktop Attendant console with accessibility options for blind or low-vision users
- **Management of urban line terminations:**
  - analogue (ETSI TBR-21)
  - ISDN PRA (ETSI TBR-4, 300 102, 300 125)
  - ISDN (BRA) (ETSI TBR-3, 300 102, 300 125)
- **Management of private network terminations:**
  - analogue E&M
  - digital QSig (2Mbit/s) (ETS 300 011/ 170/ 171/ 172/ 173/ 237/ 238/ 260/ 261)
- **Legacy Class 5 Caller ID telephone services (ETS 300 778-1)**
- **ISDN supplementary services:**
  - malicious call subscriber identification (ETS 300 128 / 129/130)
  - calling / connected line identification (ETS 300 089/ 091/ 092 / ETS 300 094/ 096/ 097)
  - Calling / Connected Line Identification Restriction (ETS 300 090/ 091/ 093/ 095/ 096/ 098)
  - hold and retrive (ETS 300196-1)
  - three-party conference (ETS 300 188-1)
  - taxation according to the “SIP pilot” and ETSI-DSS1 standards
  - remote reading of costs
- **QSIG services:**
  - basic call (ISO / IEC 11572)
  - name identification (ISO IEC 13868)
  - diversion (ISO / IEC 13673)
  - call transfer (ISO / IEC13869)
  - call offer (ISO / IEC 14843)
  - call completion on busy subscriber (ISO / IEC13870)
  - call completion on no reply (ISO / IEC 13870)
  - path replacement (ISO/IEC 13874)
  - advice of charge (ISO/IEC DIS15050)
  - message waiting indication

TELEPHONE SERVICES FOR SIP USERS

- Calling Line Identification Presentation/Restriction (CLIP/CLIR)
- Connected Line Identification Presentation/Restriction (COLP/COLR)
- Call holding and retrieval (RFC 3261)
- Call forwarding (RFC 3261)
- Call transfer (RFC 3515/3891)
- Conference (DRAFT 00)
- Instant messaging (partial RFC 3428)
- Video call (RFC2327)
- Multilinearity
- DTMF transmission / reception (RFC3261 / 2833)

UTILITY FOR SIP USERS

- Registration controlled by management system
- Hierarchical management of SIP users through service classes
- Controlled access to system resources:
  - outband output
  - abbreviated numbering
  - prohibited numbers
- Easy access to "trusted-host" SIP remote users
- Voice mail
- Group management in "SIP forking" mode (RFC 3261)
- Call forwarding / Call transfer

MANAGEMENT AND PROVISIONING

- IP Management:
  - online and offline configuration
  - FW upload of central units and peripherals via web
  - uploading welcome messages via web
  - configuration release management facility
  - online and offline maintenance
  - online alarm collection
  - accounting and CDR
- Web Personal Provisioning
- Multi-sites Management:
  - alarm
  - accounting
- SNMP-TRAP Service Support (RFC 1157)

SAMoffice 4000E

SAMIP® systems can perform advanced gateway functions in combination with the SELTA BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications

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SELTA
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SAMoffice 2C

SAMIP® Communication Server

SAMoffice 2C è il sistema di comunicazione compatto della famiglia SAMIP® in grado di soddisfare qualsiasi esigenza di aziende e uffici sino a 96 interni telefonici. SAMoffice 2C è pensato per le necessità delle aziende più attente alle innovazioni tecnologiche, come l'utilizzo di linee VoIP, di strumenti di convergenza fisso-mobile, di soluzioni che prevedono integrazioni CTI tra telefono e PC con l'obiettivo di contribuire alla velocizzazione dei processi aziendali (amministrativi, commerciali e/o produttivi). SAMoffice 2C è il compagno ideale per qualsiasi azienda per comunicare con la massima produttività, efficienza e flessibilità.

CARATTERISTICHE PRINCIPALI

SAMoffice 2C può essere utilizzato in modalità Full-IP o ibrida IP-TDM per rispondere cosi' a specifiche esigenze, presenti e/o future, di ogni tipologia di azienda. **SAMoffice 2C** è stato progettato per connettersi, in maniera nativa, a qualsiasi tipologia di linea telefonica tradizionale (ISDN o PSTN) così come a linee urbane VoIP fornite da ISP e carrier SIP. Le risorse VoIP integrate possono essere utilizzate per collegare terminali IP Selta della serie NETfon Bluelight e 51.xx oppure per telefoni a standard SIP o client SIP installati su PC fissi, notebook, smartphone o tablet collegati tramite rete WI-FI. La capacità dei sistemi **SAMoffice 2C** di collegarsi a qualsiasi rete o apparato di telecomunicazione sia IP che TDM, anche in modalità gateway, nonchè la disponibilità di un ampio numero di interfacce interne, fanno di **SAMoffice 2C** il compagno ideale per comunicare con qualsiasi strumento. Ogni sistema **SAMoffice 2C** è fornito equipaggiato con:

- Risorse VoIP attive per l'utilizzo di terminali IP o linee urbane VoIP
- Licenza TAPI 1st Party, per l'integrazione con le rubriche di MS Outlook e/o l'utilizzo di applicativi esterni su PC
- Licenze Personal Mobility
- Licenze Inbound Routing per instradare le chiamate in ingresso
- Licenze Click-2-dial per chiamare da rubrica via web browser
- Risponditore integrato (4 annunci sincronizzati sino a 32 sec. cad.)

La funzionalità Integrated Messaging rende disponibile il Voice Mail / Posto Operatore Automatico 2 porte per 32 caselle vocali e 30 minuti di registrazione di base. Tramite il P.O. Automatico e poi possibile riconoscere i fax in arrivo e far transitare sulle linee urbane attestate su SAMoffice 2C le chiamate esterne. Grazie alla funzione "Voice-2-Mail Gateway" gli utenti del Voice Mail possono ricevere il messaggio registrato direttamente nella propria casella email. I sistemi SAMIP® possono svolgere funzioni evolute di gateway in abbinamento alla soluzione **SELTA BRAVO®**, in grado di abilitare scenari di Virtual-PBX e Comunicazione Unificata, oltre ad applicazioni di Smart Working.

FUNZIONI PRINCIPALI

VoIP
Consente l'utilizzo immediato di risorse dedicate per il networking e per la telefonia IP, oppure per il contenimento dei costi grazie al collegamento con i Carrier SIP.

Affidabile
Interamente progettato e costruito in Italia secondo i più avanzati standard industriali.

Completo
Dispone di un'ampia gamma di servizi telefonici per le esigenze di qualunque utente, fisso o mobile.

Multi-sede
Consente la distribuzione geografica delle comunicazioni aziendali tramite i collegamenti IP o TDM Q.Sig con filiali, agenti, personale mobile e così via.

Aperto
Il sistema si presta all'integrazione con le applicazioni aziendali e con l'infrastruttura IT per massimizzare il ritorno dell'investimento tecnologico.

Gestione semplificata
L'amministratore può gestire il sistema via web o tramite client dedicato.

CAPACITÀ DEL SISTEMA

Linee esterne
- fino a 16 porte ISDN BRI T0/S0
- fino a 32 porte FX0 (linee trunk analogiche)
- fino a 64 porte FXS
- fino a 2 porte ISDN PRI E1/T1

Interni telefonici
- fino a 96 interni SELTA IP 51xx Series
- fino a 96 interni IP NETfon Bluelight
- fino a 96 interni IP SIP
- fino a 56 interni analogici

Risorse DSP
- fino a 72 canali VoIP con unita' DSP Farm

SPECIFICHE TECNICHE

Dimensioni e peso
Subrack (mm): H.45 x L.440 x P.400; 5 Kg

Montaggio
in rack dati 19'' (1U) o a parete

Connettori
- Collegamento con la rete LAN: 10/100 Base T Ethernet port switch, IEEE 802.3/802.3u, AutoSense, Automatic cable crossover (MDI/DI-X), PoE 802.3af
- Porta AUX per ingressi e uscite verso il campo (2 DIN, uscita relè) per collegamento allarmi, pulsanti, ripetitori di chiamata.
- Porta COM-RS232 RJ45 per documentazione adebiti
- Porta COM-RS232 CONSOLE RJ45

Alimentazione
- Alimentatore esterno AC/DC
- AC INPUT : 100-250 Vca 47/63 Hz
- AC OUTPUT : - 48V DC : 1A

Consumi
Dati stimati per il sistema base:
- SAMoffice2C max.: 60 W

DATI DI SISTEMA

BHCC
Fino a 30.000 Busy Hour Call Completion

Unità Centrale
Processore principale Intel Xscale, matrice di commutazione TDM da 4096 canali e 16 fili PCM, processore ausiliario ARM

Sistema Operativo
Linux

CERTIFICAZIONI
- Marcatura CE (direttiva 1999 / 05/EC della Comunità Europea: ETSI EN 300 386; EN 55022; EN 55024; EN 61000-3-2; EN 61000-3-3; EN 60950)
- Interoperabilità su rete Hyperway di TIM (rete di centrali SAMIP)
- Interfaccia PSTN (ETSI TBR-21)
- Interfaccia rete pubblica ISDN-PRA (ETSI TBR-4; ETS 300102; ETS 300125)
- Interfaccia Rete Pubblica ISDN-BRA (ETSI TBR-3; ETS 300102; ETS 300125)
- Caller ID (ETS 300 778-1)
- Test ambientali (ETS EN 300 019 class 3.2 in use) eseguiti presso laboratorio interno

APPLICAZIONI E CTI

- TAPI 1st party (Microsoft TAPI 2.2)
- TAPI 3rd party (Microsoft TAPI 2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - ACD multicanale
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - click to talk (VoIP)
  - call me back
  - co-browsing
- CRM oriented
- Reporting statistico
- Call Flow Editor:
  - VM/AA integrato
  - facility UM
- Unified Messaging:
  - voice2email
  - fax2email
  - sms2email
- Servizi Hotel integrati:
  - su telefono e P.O.
  - web hotel su browser
  - link con PMS esterni

CARATTERISTICHE VoIP

- Segnalazioni
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Codifiche vocali
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Jitter Buffer configurabile
- QoS tramite DiffServ (RFC 2474/2597/2598)
- QoS tramite TOS (RFC791)
- Supporto gatekeeper H.323 (ITU-T)
- Backup su linee urbane
- Music on hold

VoIP AND ToIP SERVICES

- SIP gateway
  - terminali analogici/ digitali
  - linee analogiche/ISDN (BRI/PRI)
  - QSIG networking
  - S0 Data
- H.323 gateway (ITU-T H.225.0)
- SIP networking con protocollo SIP-T (RFC 3204)
- Fax T38 (ITU-T)
- Supporto terminali SIP standard:
  - hardphone
  - softphone
- Supporto terminali Wi-Fi SIP standard (IEEE 802.11)
- Supporto terminali SIP ibridi Wi-Fi - 2/3 g (IEEE 802.11/ITU-T IMT-2000)
- Supporto terminali H.323 standard (chiamata base) (ITU-T)
- Terminali IP proprietari per servizi KTS
  - hardphone (NETfon)
- Connessione via internet alla LAN aziendale con telefono NETfon
  - telelavoro
  - sede periferica
- Power-over-Ethernet (PoE) (IEEE 802.3af)

	BASE	MAX
UTENTI IP	16	96
UTENTI TDM	16	64
SLOT DISPONIBILE	2	4
MODULI ESTENSIONE		1

* Equipaggiamento massimo non raggiungibile contemporaneamente

SERVIZI LEGACY PBX

- **Gestione terminazioni di utente**
  - analogiche (CEI 103, TBR 21)
  - digitali proprietarie (famiglia SAEfon CL) (CEI 103)
  - ISDN su interfaccia S0 (ETSI TBR-3)
- **Posto Operatore (CEI 103)**
  - consolle supervisore
  - consolle operatore da tavolo su PC (Microsoft TAPI 2.2)
  - Console operatore desktop con opzioni di accessibilità per utenti non vedenti o ipovedenti
- **Gestione terminazioni di linea urbana**
  - analogica (ETSI TBR-21)
  - ISDN PRA (ETSI TBR-4, 300 102, 300 125)
  - ISDN (BRA) (ETSI TBR-3, 300 102, 300 125)
- **Gestione terminazioni di rete privata**
  - analogica E&M
  - digitale QSig (2Mbit/s) (ETS 300 011/ 170/ 171/ 172/ 173/ 237/ 238/ 260/ 261)
- **Servizi telefonici ID chiamante classe 5 legacy (ETS 300 778-1)**
- **Servizi supplementari ISDN**
  - identificazione abbonato disturbatore (ETS 300 128 / 129/130)
  - calling / connected Line Identification (ETS 300 089/091/092) (ETS 300 094/096/097)
  - Restrizione identificativo linea chiamante/linea connessa (ETS 300 090/ 091/ 093/ 095/ 096/ 098)
  - hold e retrive (ETS 300196-1)
  - conferenza a tre (ETS 300 188-1)
  - tassazione a standard "pilota SIP" ed ETSI-DSS1
  - telelettura costi
- **Servizi QSIG**
  - basic call (ISO/IEC 11572)
  - name identification (ISO IEC 13868)
  - diversion (ISO/IEC 13673)
  - call transfer (ISO/IEC13869)
  - call offer (ISO/IEC 14843)
  - call completion on busy subscriber (ISO/IEC13870)
  - call completion on no reply (ISO/IEC 13870)
  - path replacement (ISO/IEC 13874)
  - advice of charge (ISO/IEC DIS15050)
  - message waiting indication

SERVIZI TELEFONICI PER UTENZA SIP

- Presentazione/restrizione dell'identificativo della linea chiamante (CLIP/CLIR)
- Presentazione/restrizione dell'identificazione della linea connessa (COLP / COLR)
- Chiamata in attesa e recupero (RFC 3261)
- Inoltro chiamata (RFC 3261)
- Trasferimento di chiamata (RFC 3515/3891)
- Conferenza (DRAFT 00)
- Messaggistica istantanea (RFC 3428 partziale)
- Videocall (RFC2327)
- Multilinearità
- DTMF trasmissione/ricezione (RFC3261/2833)

UTILITY PER UTENZA SIP

- Registrazione controllata da sistema di gestione
- Gestione gerarchica utenza SIP tramite classi di servizio
- Accesso controllato alle risorse di sistema
  - uscita in outband
  - numerazione abbreviata
  - numeri vietati
- Accesso agevolato a utenza remota SIP di tipo "trusted-host"
- Voice mail
- Gestione gruppi in modalità "forking SIP" (RFC 3261)
- Inoltro chiamata / trasferimento di chiamata

MANAGEMENT E PROVISIONING

- IP Management
  - configurazione online e offline
  - caricare messaggi di benvenuto via web
  - caricamento annunci di cortesia via web
  - facility di gestione delle release di configurazione
  - maintenance online e offline
  - raccolta allarmi online
  - accounting e CDR
- Web Personal Provisioning
- Multi-site Management:
  - alarm
  - accounting
- Supporto Servizio SNMP-TRAP (RFC 1157)

SAMIP® systems can perform advanced gateway functions in combination with the SELTA BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications

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