Avenida de la Albufera, 323, pl. 2, of. 13
28031 Madrid (Spain)
Phone: +34 91 803 9251
E-mail:wirelessmundi@wirelessmundi.com

Web:http://www.wirelessmundi.com

WORLD COMMS

Carrier & Service Provider

FEATURES & TECHNICAL SPECIFICATIONS

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INTRODUCTION

The specifications, main functions and usage scenarios of the modules of the COMMS MUNDI solution are described below.

POSSIBLE USES

- Session Border Controller (SBC)
- Operator softswitch
- Business switchboard (IP-PBX)
- Multi-company virtual PBX server (Multi-tenant PBX)
- transcoding device
- Protocol conversion gateway (ISDN->SIP...)

COMPATIBILITY

- Switchboards of any kind (Traditional, VoIP...)
- Analog gateways to VoIP (ATA)
- SIP phones of any brand and model.
- SIP intercoms (with or without video).
- SIP public address systems.
- Softphones on computers or mobile devices
- Integration with databases of external systems
    - Billing services (pre/post payment).
    - Enterprise CRM software.
    - CTI integration (call center, TAPI...).
    - Automatic marking systems.

LIMITATIONS

- Unlimited number of extensions.
- Unlimited number of lines and DID.
- Unlimited number of simultaneous calls.
- Unlimited number of call routes and number lists.

BASIC CALLING FUNCTIONS

- Caller ID (CallerID).
- Caller ID (CalleeID).
- Conference to three (3-way calling).
- Call forwarding if busy.
- Call forwarding if you do not answer.
- Unconditional call forwarding.
- Do not disturb (Do Not Disturb).
- Call retention (Hold).
- Transfer of attended / supervised calls.
- Blind / unmonitored call transfer.
- Music on hold and transfer.
- Mailbox Message Indication (MWI)
- Custom tones (custom ringback)
- Groups of simultaneous extensions (Ring groups).
- Groups of sequential extensions (Hunt groups).

SECURITY

- SIP authentication support
- Voice encryption by SRTP and ZRTP
- Configurable maximum call duration per route
- Configurable number of simultaneous calls per route
- Number of calls per period of time configurable by route
- Lists of unauthorized origins or destinations
- ACLs by IPs / ranges /...

MONITORING AND RECORDS

- Monitoring of calls in progress.
- Status monitoring of lines and local accounts
- Registration and lists of incoming and outgoing calls (CDR).
    - nanosecond granularity
    - Exportable to external ODBC database.
    - Format and configurable variables

CALL ROUTING

- Shared or individual outgoing routes by telephone, virtual switchboard...
- Shared incoming routes or by line.
- List management.
    - Black lists (numbers with prohibited access).
    - White lists (authorized numbers...).
    - Correspondence lists (short numbering, aliases...)
- Routing according to call parameters
    - Destination of the call (destination number, geographical number contacted (DiD...)
    - Origin of the call (line or account, caller ID...)
    - Advanced SIP parameters (X-Headers, etc.)
- Routing according to external parameters
    - Date and times (working hours, holidays, etc...)
    - ODBC Database Query
    - Predefined variables Day / night mode
- route actions
    - Call to fixed local destination or using call parameters (typed destination, prefixes...)
    - Call to fixed remote destination or using call parameters (typed destination, prefixes...)
    - Automatic call failover to other destinations
    - Simultaneous calls to multiple destinations
    - Running applications or local voice services.

APPLICATIONS / VOICE SERVICES

- Automatic detection of incoming faxes and forwarding or receiving if detected.
- DTMF digit authentication (security for special calls...)
- Answering machine / voice mail
- fax reception
- Recording of incoming and outgoing calls on demand.
- Recording of incoming and outgoing calls permanent by parameters.
- Audio playback (wav/mp3 formats)
- Capture of calls not established (pickup).
    - Direct extension capture.
    - Capture group of extensions.
- Interception of established calls.
    - Determined extent intercept.

    - Extension group intercept.
- Listening to established calls.
    - Listening to a certain extension.
    - Listening to group of extensions.
- Digital Operator (IVR, hierarchical interactive menus)
    - Multiple hierarchical menus
    - recordable instructions
- Conference rooms (2 or more terminals simultaneously).
- Automatic call distribution (ACD, queuing solution for Call Centers).
- Generation of outgoing calls (Auto-Dialer, automated surveys...).
- DISA system: a person outside the office can make calls through the switchboard.
- Automatic redial (Callback, connect with a phone with a destination after missed call)
- Custom programming of new functionalities!

CENTRAL DIRECTORY OF ACCOUNTS AND GROUPS WITH MULTIPLE DOMAINS / VIRTUAL SWITCHBOARDS

- Option of voice mailboxes (individual or shared).
    - Password protection.
    - Configurable mailbox response delay.
    - Consultation and management through the telephone terminal.
    - Consultation and management through web interface.
    - Integration with email (voicemail to email).
- Option of fax mailboxes (individual or shared).
    - Send and receive faxes from the system itself.
    - Consultation and management through web interface.
    - Integration with email (fax to email).
- Option of recording mailboxes (individual or shared).
    - Consultation and management via web interface
    - Email integration (recordings to email)

INTERCONNECTION / STANDARDS AND PROTOCOLS

- Analog operator lines (FXS) and user lines (FXO)
    - Loop Start (LS) and Kewl Start (KS) signaling
    - Caller ID detection: FSK, DTMF, ETSI (before or after ring)
    - Hardware or software amplification and echo cancellation
- ISDN operator digital lines (ISDN-BRI TE mode) and user lines (ISDN-BRI TE mode)
    - Point to Point (PTP) or Point to Multipoint (PTMP)
    - Hardware or software amplification and echo cancellation
- Operator PRI digital lines (ISDN-PRI E1/T1/J1 NET mode) and user (ISDN-PRI E1/T1/J1 CPE mode)
    - CSS or CAS frames and HDB3 or AMI encoding with optional CRC4
    - ISDN or MFC/R2 signaling
    - Connection with EuroISDN switches, MFC/R2, National, DMS100, 4ESS, 5ESS, QSIG
    - Hardware or software amplification and echo cancellation
- Voice lines over IP (VoIP protocol SIP)
    - SIP Signaling v2.0 (RFC 3261, RFC 3263, RFC 3325) over UDP, TCP
    - Presence / Chat (SIP/SIMPLE)
    - Voice over RTP, SRTP, ZRTP
- codecs
    - standards: G.711 (a-law/mu-law), G.726 AAL2, DVI4 (ADPCM)
    - High definition (HD) codecs: G.722, G.722.1 (Siren 7), G.722.1C (Siren 14), CELT, OPUS, SILK (Skype codec)
    - Low Bitrate Codecs: Speex, iLBC, GSM, BV16/32 (Broadvoice), Codec2, LPC10

    - Video codecs (passthrough): H.261, H.263, H.263-1998, H.263-2000, H.264, MP4, Theora
- Fax over IP: Audio T.30 or T.38 IP Fax
- DTMF digits: RFC-2833, Inband, SIP INFO

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