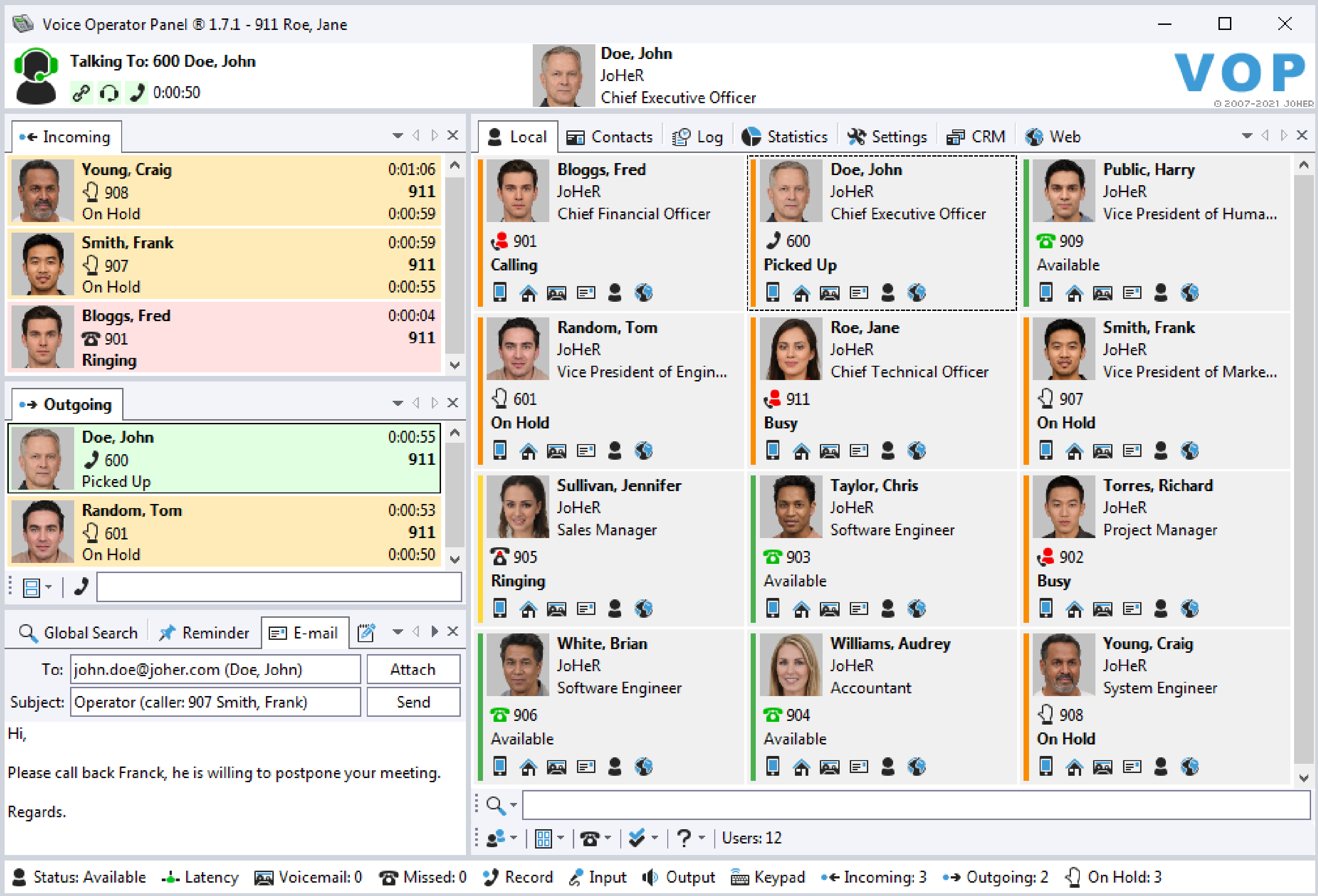
Voice Operator Panel® Datasheet



Voice Operator Panel is a professional softphone and attendant console for operators and receptionists.

The application can be used as a standalone softphone or tethered to any analog or VoIP hard phone to handle calls.

The user can pick up incoming calls, put them on/off hold, initiate unattended transfers or call users and initiate attended transfers.

Transfers can be initiated through a contextual menu, using configurable keyboard shortcuts or using intuitive drag and drop.

Multiple users directories are supported, they can be imported from CSV or vCard files, loaded from LDAP servers or from Outlook contacts.

Each directory contains a list of users.

Each user has a picture and information such as Company, Office, Department, Title.

Each user has multiple contacts such as Phone, Mobile, Home, Voicemail, E-mail, XMPP, Calendar, Web.

Each directory supports enhanced search by Name, First Name, Company, Office, Department, Title, Phone, Mobile, Home, E-mail, XMPP.

Each directory supports global default contact selection and contact/information filtering.

Each directory supports user presence using XMPP when available.

Each phone contact supports phone presence when available. (phone icon changes when phone is ringing or busy and calls details are shown)

Each phone contact supports messaging when available.

Each phone contact supports call interception when available.

The application provides a detailed call log supporting filtering and search.

The status bar offers enhanced features such as user status change, latency tracker, voicemail indicator, missed incoming calls indicator, call recording, input sound volume/muting/effects, output sound volume/speaker, DTMF keypad and call statistics.

Voice Operator Panel speeds up the call processing:

Each call has an icon showing the status (ringing, on hold, picked up) and a background color showing the processing priority (red > orange > green).

For each call on hold, the duration is shown and turns to bold/red when a configurable value is reached.

The user can tag incoming calls adding a custom line to ease the processing.

Callers are identified from directories. Anonymous calls or unidentified callers can be automatically rejected.

After a configurable delay, incoming ringing calls can be automatically forwarded, rejected or answered and put on hold.

The number of concurrent incoming calls can also be limited by the user. Extra calls are forwarded or rejected.

Within the application, the user can quickly load CRM callers records using the built-in web browser or send e-mails using the built-in e-mailer.

Voice Operator Panel is ideal for ITSP (Internet Telephony Service Providers):

The configuration file supports more than 200 parameters giving ITSP full control over the end users application behaviour.

The user settings window is built from the configuration file allowing ITSP to select which settings are offered to end users.

The Automatic Provisioning System allows ITSP to easily handle end users application binary and configuration updates remotely.

Using the built-in web browser, the application can display an online web settings page managed by the ITSP.

The hard phone support allows ITSP to easily bring end users to switch from analog PBX to IP PBX or IP Centrex.

ITSP can select among three operating modes enhanced for dedicated voice network or mixed data/voice network.

A multi-level and multi-layer logging system with syslog support allows easy remote debugging.

Multiple languages are supported, the translations can be easily modified by the ITSP to better fit with its environment.

Voice Operator Panel has been fully validated with Metaswitch, Broadsoft, Cisco, Cirpack, Vodia, 3CX, Asterisk, sipXecs, FreeSWITCH, Kamailio.

Technical Specification

Computer: PC, Processor Intel Pentium IV or better, 256MB RAM free or more, 50MB disk space free or more.

Operating System: Windows Vista, 7, 8, 10. (with .NET 4.0 or later framework installed)

Protocol: UDP/TCP/TLS SIP (RFC 3261, 3263, 3264, 3581, 3891, 4028, 4488, 4916, 5806) SDP (RFC 4566, 4568) RTP (RFC 3550) SRTP (RFC 3711)

Codec: G.722, G.711, G.729.

DTMF: In-band, Out-band (RFC 2833)

User Presence & Messaging: XMPP (RFC 3920, 3921)

Phone Presence: SIP SUBSCRIBE/NOTIFY (RFC 3265, 4662) Dialog (RFC 4235) or Presence (RFC 3856) PIDF/XML (RFC 3863) RPIDF (RFC 4480)

Own Presence: SIP SUBSCRIBE/NOTIFY (RFC 3265) Dialog (RFC 4235) or SIP PUBLISH (RFC 3903) Presence (RFC 3856) PIDF/XML (RFC 3863) RPIDF (RFC 4480)

Phone Messaging: SIP MESSAGE (RFC 3428)

Voicemail: SIP SUBSCRIBE/NOTIFY (RFC 3265) Message-summary (RFC 3842)

Call Transfer: SIP REFER (RFC 3515)

USB Devices: Headset, Handset, Speakerphone, Light/Ringer.

Outlook Support: 2000, 2002, 2003, 2007, 2010, 2013, 2016, Office365.

Languages: English, French, Spanish, German, Italian, Dutch, Danish, Portuguese, Polish, Turkish, Russian.

Validated with: Metaswitch, Broadsoft, Cisco, Cirpack, Communigate, Enswitch, Vodia, 3CX, Asterisk, sipXecs, FreeSWITCH, Kamailio, OpenSIPS.