ATTIMO®

CLOUD PBX MULTISERVICES

ATTIMO MULTISERVICES is a Cloud Corporate Communication Platform that provides advanced IP Telephony and Mobility features for different needs and user profiles from small, medium and large enterprises with unlimited number of users. With public cloud hosting in Brazil, it follows all security, availability and confidentiality standards combined with the provision of highly reliable and simple access to communication services through any environment. Developed in line with the latest market trends such as WebRTC and Opus, **ATTIMO** provides flexible services that meet the requirements of new workspace.

Call Features

- Compatible with standard SIP Phones.
- User Language Telephony Interface: English, Portuguese-BR, Spanish, French and Italian.
- WebRTC Softphone.
- Windows, Mac, iOS and Android Soft Clients
- Multi-device (5 simultaneously)
- Multi Boss/ Multi Secretary + Call List.
- Busy Lamp Field (BLF) Support.
- Free Seating
- Call Back.
- Call Flow Control.
- Call Forward (busy, no answer, unconditional).
- Call Monitoring.
- Call Pickup.
- Call Parking.
- Call Recording (Record, Search, Listening).
- Call Recording Audit and File Download.
- Call Screening.
- Call Spy.
- Call Transfer (blind, attended).
- Call Waiting.
- Callback Services.
- Caller-ID.
- Call Consultation (External and Internal)
- Connected Number ID.
- 3 Party Conference Call
- 30 Party conference bridge
- DISA (Direct Inward System Access).
- DND (Do Not Disturb).
- Direct Inward Dial Numbers (DIDs).
- Find Me/Follow Me.
- Hunt/Ring Groups.
- Hot-line/ Warm-line.
- Music on Hold.
- Outbound/Inbound Routes.
- PIN Sets.
- Dial restriction per set
- Incoming Call restriction based on Caller ID.
- Outgoing Call restriction based on Called ID.
- Blacklist (PSTN, LOCAL, LONG DISTANCE LDN).
- Blacklist (Collect Call, 0300, 0500, 0900).
- DID Blockage.
- Pickup Groups.
- Personal Assistance (IVR).
- Personal Recording/Notes.
- Remote Substitution.
- Remote Call Pickup.
- Route by Caller ID.
- Speed Dials.
- Video Calling.
- Wake-up Calls.
- Multiple Devices per User.
- Personal Call Log.
- Personal Extension Settings.
- Personal IVR.
- UC (Chat, Video and Sharing).
- Alert Message Broadcast on SIP Sets
- Local SIP Survivability.

Protocols, Codecs and RFCs

- PJSIP., SIP, SIP connect and SIP register.
- IAX2.
- PRI/T1/E1/R2.
- Analog/FXS/FXO.
- ISDN.
- GSM.
- g729, g719, g722, g723.
- lpc10, slin.
- g711 A-law, g711 ulaw.
- siren14, siren7.
- g726, g726aal2.
- speex/16/32.
- adpcm, testlaw.
- gsm, lilbx, vp8, vp9.
- h261, h264, h263p, h263
- Opus
- H.264 (QCIF, CIF, VGA e HD)
- IPV4 and IPV6.
- IP Address and DNS.
- SIP session-aware NAT/PAT for RTP/SRTP media.
- NTP.
- SSH2 and HTTPS.
- DTMF
- SNMP, SNMP V2.

Security

- Built-in firewall.
- End to end Call Encryption including on Call Transfer and Conferences.
- Call encryption (SIP TLS, sRTP, DTLS) for SIP Trunks, SIP Devices and Softphone.
- AES 128 and 256.
- RTP/SRTP mediation.
- DMZ and NAT operation
- Authorization codes.
- HMA-SHA-1
- Phone Lock.
- Intrusion detection and -blocking (Fail2ban).
- Limiting or blocking of outbound calls.
- Password strength indicator.
- PIN-protected outbound calls.
- User permission management.
- Secure password auto-generation.
- Weak password report.
- HTTPS.
- OpenVPN.

Voice Mail

- Visual Indicator for Message Waiting.
- Stutter Dial tone for Message Waiting.
- Voicemail to email.
- Voicemail Groups.
- Voicemail Broadcast.
- Web Voicemail Interface.
- Direct dial to operator.

Multimedia Conference Services

- Up to 1.000 users per room
- Unlimited rooms
- Multiple Presenters
- WebRTC
- SIP Dial-in
- Audio, Video, Chat
- Conference Recording
- Screen Sharing
- Video and Audio Sharing
- Customized Background
- Noise Canceling
- Raise Hand
- White Board
- Pooling
- Video Mosaic
- Multi-party Continuous Presence
- Presenter statistics

Availability and Hosting

- 99,5% Availability
- 24 hours x 7 days
- Multi-DC Sao Paulo Zone
- Huawei Cloud ISO 27001
- ODATA Tier 3 Certified DC
- Multi-Carrier Connection

ATTIMO®

CLOUD MULTISERVICES

In addition to Cloud Corporate Communication Services, **ATTIMO MULTISERVICES** offers Call Center Services that is designed to provide exceptional customer service through efficient and effective communication. Equipped with advanced technology, Attimo Call Center ensures that every customer interaction is handled with the utmost care and professionalism.

Call Recording
- Web management GUI.
- Administrator Language Interface: English, Portuguese-BR, Spanish, French and Italian.
- Centralized recording for single or multi-site topologies.
- Multi domain and customized views.
- Search interface with caller id, called extension, date, time and call period.
- Extensions Groups Recording
- Multi Level Admin Access Profile
- Up to 1 year storage on Cloud.
- Recorded files export to On-Premises environment.
- Recorded files format .mp3 and .wav.

Agent Features
- Agent login/log out.
- Free Seating.
- Pause
- Withdraw
- Queue ID
- Caller ID
- Transfer to IVR / Agent/ Supervisor
- Multi Tab

Call Center
- Agent login/log out.
- Free Seating.
- Barge.
- Call monitoring.
- Call queues.
- Call recording.
- Caller Name Lookup.
- CDR (Call Details Record).
- Conferencing (on-the-fly).
- Customer account codes.
- Hot-desking.
- Hunting groups.
- IVR Multilevel
- Auto-attendants.
- Satisfaction survey
- TTS and ASR
- Listen to agent.
- Pick-up groups.
- Advanced Routing
- Skill base routing
- Time-based routing.
- Queue priorities.
- Queue VIP list.
- Ring group strategies.
- Reporting.
- Whisper to agent.
- Dialler/ Outbound Calls

Supervision Features
- Real time statistics
- Alarms and reports
- Queue Status.
- Agents' status
- Historical Reports
- IVR Flow

Reports
- Inbound Calls
  - Per Hour
  - Per day.
  - Per week.
  - Per month.
  - By Queue.
  - By Group.
  - By Agent.
- Outbound Calls
  - Per Hour
  - Per day.
  - Per week.
  - Per month.
  - By Group.
  - By Agent.
- Answered Calls
  - Per Queue.
  - Per Group.
  - Per Agent.
  - Transferred.
- No answered Calls
  - Per Queue.
  - Per Group.
  - Per Agent.
- Call duration
  - Total call time.
  - Total call time rate.
  - Average call time.
  - Waiting time.
  - Average waiting time.
  - Maximum waiting time.
- Queue Waiting
  - Expected waiting time
  - Per Queue
- Ring time
  - Ring time rate.
  - Maximum ring time.
- Agents
  - Availability
  - Pauses
  - Withdraw
  - Answered calls
  - No answered calls
  - Disconnected Calls
  - Disconnect by Caller
  - Transferred
- Service Level.
- Cause of disconnection.
- Abandonment rate.
- Abandonment by caller.
- Customized Reports.

An Advanced Cloud PBX must be composed by Advanced Centralized Administration Services that offers a comprehensive solution for managing your business's telephony needs with ease and efficiency. Designed to streamline communication and enhance productivity, **ATTIMO MUILTISERVICES** provides a robust platform for handling all aspects of your Cloud PBX system.

Administration
- Web management GUI.
- Multi level access profile.
- Administration Logs.
- Centralized administration for single or multi-site topologies.
- Multi domain and customized views.
- User configuration.
- Trunk Configuration.
- User profile configuration.
- Centralized SIP Phone Mass provisioning.
- Auto Provisioning.
- Import and Export users data base.
- SIP Sets Real time supervision (Busy, Free, Up and Down).
- Alarms with different levels.
- Traffic Reports.
- Billing Reports.
- Quality VoIP and QoS Reports (MoS).
- Scheduled Reports