Juphoon

Datasheet

VoIP SDK

We are dedicated to providing Usable, Simple and Elegant (USE) communication products and solutions.

Deliver most comfortable video communication experience on mobile networks

Mobile VoIP users often suffer from acoustic echo, latency and distortion issues, due to wireless networks are usually unstable and vulnerable. Thus, enhancing video call quality is the key to improve user experience.

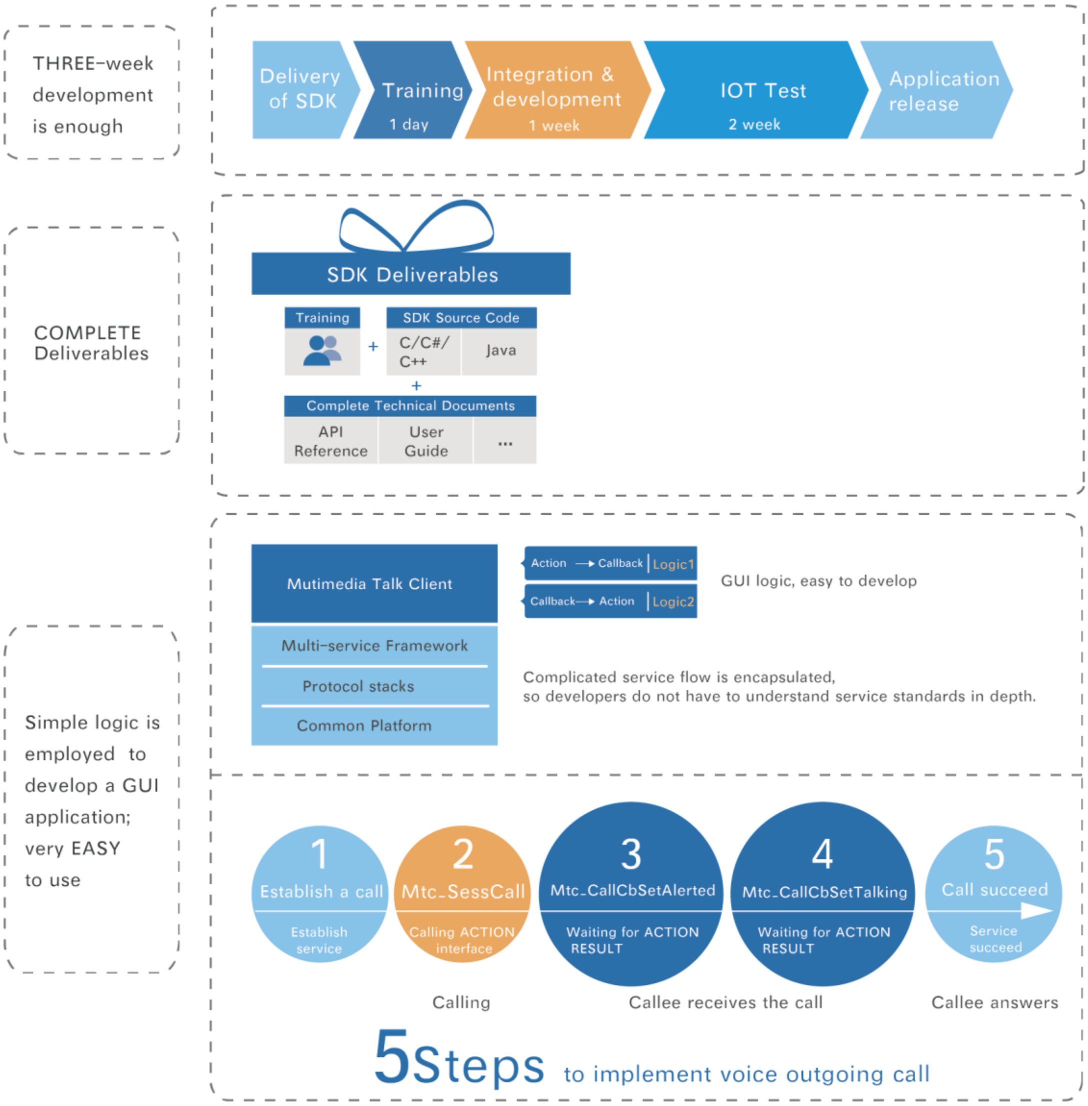
Juphoon VoIP SDK has optimized voice and video quality for mobile networks and diversified devices. It employs our advanced SPo (Sweet Point) control, bandwidth efficient mode, auto-adaptive negotiation and MDM (Media Device Management) technology to provide smooth and clear video call experience, even over unreliable networks. Apps developed with Juphoon VoIP SDK will work better on mobile networks than the like does. It can easily realize 720p/ 1080p HD video experience on PCs and VGA (640x480) output on smartphones. And it ensures smooth experience even when packet loss rate hits 30%.

- HD voice and video experience
- Reduce video call data usage
- Adapt video quality to device capabilities
- Fast delivery and easy to use
- Excellent compatibility and interoperability

Core Features

Easy to use

Juphoon VoIP SDK is simple and user friendly. Its user interfaces are named in an intuitive way and can be directly integrated with GUI logic. Thus it is quite easy to use and only a few couple lines of code are needed to realize a function. The SDK is delivered with professional technical documents and support which can help you launch competitive client products asap.



Excellent compatibility and interoperability

Supported OS

Windows (XP, 7, 8), Android(2.3-4.x), iOS(5.x-7.x), Linux, etc.

Supported devices

Android mobile, Android tablet, PC, and iPhone (4, 4S, 5, 5s, 5c), iPad (2, 3, 4, mini) , etc.

Interoperability

It can work with IP-PBX, Conference, IMS（MMTel）, Softswitch and IP Phone. And it has passed interoperability tests with servers of Ericsson, Huawei, Alcatel-Lucent, NSN, ZTE, Cisco, etc. It also can work with popular IP-PBX, IMS, Softswitch, SIP and SBS network devices in the market.

Premium voice and video quality

A superior multimedia engine is embedded to support HD voice and video call and ensure the best possible calling experience for every user, though they may use different devices and/or call over unreliable or heterogeneous wireless networks.

Transfer smaller images to devices with lower resolution and/or performance to achieve smooth experience.

Transfer larger images to capable devices for outstanding HD video experience.

Clear and smooth, even on unstable networks

SPo enables perfect adaptation of video transfer to bandwidth fluctuation, helping deliver clear and smooth video experience, even over unstable and/or heterogeneous networks.

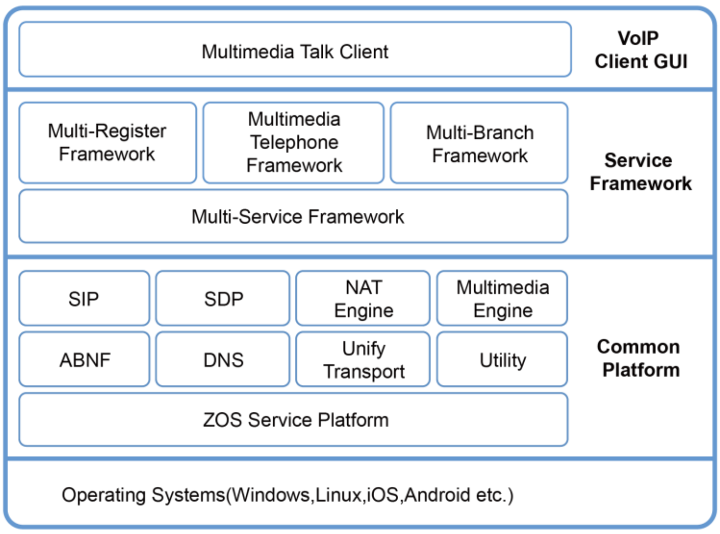
Reduce video call data usage

Bandwidth efficient mode is applied to reduce video call data usage by 30% up to 90% off, without a perceptible decrease in video quality.

Adapt video quality to device capabilities

MDM can help optimize video quality to match device capabilities automatically, thus delivering best possible video quality on all supported devices and reducing deployment costs.

Architecture



Specification

Voice and Video call

- Basic Call
- Call Hold
- Call Transfer
- Call Forward
- Call Waiting
- Caller-ID Display
- Multi-party Conference
- Custom Ring
- TEL URI
- DTMF (InBand, OutBand, INFO)
- Mute
- Early Media
- Session Timer

Network

- IPv4
- IPv6

3GPP MMTel

- Originating Identification presentation (OIP)
- Originating Identification restriction (OIR)
- Terminating Identification presentation (TIP)
- Terminating Identification restriction (TIR)
- Communication Forwarding Unconditional (CFU)
- Communication Forwarding on not Logged (CFNL)
- Communication Forwarding on Busy (CFB)
- Communication Forwarding on not Reachable (CFNRc)
- Communication Forwarding on No Reply (CFNR)
- Barring of All Incoming Calls (ICB)
- Barring of All Outgoing Calls (OCB)
- Barring of Outgoing International Calls(OICB)
- Barring of Outgoing International Calls – ex Home Country (OICBe)
- Barring of Incoming Calls - When Roaming (ICBr)
- Communication hold (HOLD)
- Communication Barring (CB)
- Message Waiting Indication (MWI)
- Communication Waiting (CW)
- Conference (CONF)
- Explicit Communication Transfer (ECT)
- Emergency call

Registration/ De-registration

- Initial registration
- Refresh registration
- Cancel registration
- Register Event Subscription
- MWI Event Subscription

Authentication

- SIP DIGEST Authentication
- IMS AKA Authentication

Network Protocols

- SIP
  - UDP/TCP/TLS
  - MTU apply of RFC3261
- RTP/RTCP
  - RTP Profile
  - Data Transport
  - RTCP Usage
- DNS Query
- STUN/TURN/ICE
  - NAT Traversal

Voice Media

- Codecs: G.711, G.722, G.729, iLBC, iSAC, AMR-NB/WB, Opus
- AEC/Adaptive AEC, AES
- ANS
  - NS-MIC, Noise Suppressor for Microphone
  - NS-SPEAKER, Noise Suppressor for data from network
- AGC
  - AGC-MIC, Auto Gain Control for Microphone
  - AGC-SPEAKER, Auto Gain Control for Speaker
- VAD (VAD Voice Activity Detection)
- CNG (Comfortable Noise Generation)
- PLC (Packet Loss Concealment)
- FEC, RED, ARS
- DTMF
  - Inband ITU Q.23
  - Outband RFC2833
  - IR92 3.3
- Very Fast Adaptive Jitter Buffer
- Front End Handling in IR92 3.2.7
- Voice quality diagnosis

Video Media

- Codec: H.264, H.265, H.263, VP8
- FEC (Forward error correction)
- RED (Redundancy)
- TMMBR/TMMBN
- SPo (Sweet Point Control)
  - ARS(Auto bit Rate Sensing)
  - Framerate Auto Control
  - Resolution Auto Control
- FIR(full intral frame request)
- Color Enhance
- Render
  - Render in Isolated Window
  - PiP Picture in Picture
  - External Render
- Adaptive Jitter Buffer
- Packet Lost, PLC Packet Loss Concealment
- Video quality diagnosis

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