# VoIP Engine™ SIP Reference Kits: Jump Start your Voice over IP App!

ADAPTIVE DIGITAL'S VOIP ENGINE/SIP REFERENCE KITS FOR IOS, ANDROID™, AND LINUX INCLUDE TWO POWERFUL SOFTWARE DEVELOPMENT KITS (VOIP ENGINE AND SIP SDKS) PROVIDING POWERFUL AND HIGHLY CUSTOMIZABLE SOFTWARE ENVIRONMENTS TO QUICKLY ADD VOIP, AND SIP BASED DIAL, ANSWER CALL FEATURES INTO AN APPLICATION.

Evaluate Today! The reference kits provides a jump start for creating a VoIP application. The sample application can be used for reference or modified to create a custom application. The VoIPEngine and Sip class libraries can be used in other custom applications. They are independent of the sample application.

Description | FAQs | iVoIPEngine™ for iOS | AnVoice™ for Android | LnxVoice™ Linux | x86

## DESCRIPTION

Adaptive Digital Technologies Simplifies Mobile Handset VoIP Application Development. Harness the Power of VoIP Engine: Voice Engine/SIP Software Development Kits for VoIP Application ARM®, iOS, and Android™ Developers.

Developers can add VoIP features to an existing hardware or software project or create a fully customized SIP application. The included sample SIP Phone app is a fully functioning SIP phone. The app can be configured to connect to a standard SIP server. The app can place outgoing phone calls as well as receive inbound phone calls. Furthermore, it supports peer to peer VoIP for applications that do not require SIP.

**SOFTWARE** FEATURES

### SIP Phone Application

- SIP Client Protocol
- RTP/Jitter Buffer
- SRTP
- Voice Conferencing (up to 4 users)
- G.711 mu-law and a-law with packet loss concealment
- G.729A 8 kbps speech compression
- G.722 16 kbps speech compression
- Noise Reduction
- Enhanced Acoustic Echo Cancellation – Operates on most Android handsets without customization/tuning.
- Automatic Gain Control
- Tone Generation
- Tone Relay Transmit
- Peer to Peer Operation
- Fully Configurable via GUI
- G.722.2 Vocoder (Optional)
- AMR Vocoder (Optional)

### Test Features

- Tone transmit
- Tone receive
- CSS transmit
- CSS receive
- Acoustic Delay Measurement

Launch Extranet LOG-IN | REGISTER to download free Reference Kit

Click on Tools > Demos, then Click Download Demonstrations, choose from list.

**AnVoice - VoIP Engine / SIP Reference Kit for Android** FREE Download - registration required

**LnxVoice** - VoIP Engine / SIP Reference Kit for Linux FREE Download - registration required

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VoIP Engine™
Voice over IP Engine Suite

RELATED PRODUCTS:

- VoIP Engine
- AnVoice
- iVoIPEngine
- LnxVoice
- LnxVoice demo - Sitara 335x

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*Adaptive Digital's VoIP Engine™/SIP Reference Kits can be used to accelerate the development of cutting edge VoIP applications, while delivering the best user experience for mobile clients.*

*Give us a listen!*

Adaptive Digital is the leading developer of VoIP technology. Our VoIP Engine SDK provides a powerful and highly customizable software environment to quickly add SIP based dial, answer call features in your application. Using Adaptive Digital's VoIP Engine software development kit (SDK) accelerates the development of SIP compliant voice applications. Developers can add VoIP features to an existing hardware or software project or create a fully customized SIP application.The sample SIP Phone app is a fully functioning SIP phone. The app can be configured to connect to a standard SIP server. The app can place outgoing phone calls as well as receive inbound phone calls. Furthermore, it supports peer to peer VoIP for applications that do not require SIP.

The SIP phone app makes use of underlying SIP and VoIP Engine services that are accessed through their respective APIs. Both services run autonomously.

In the case of the SIP service, it runs in its own thread but under control of the sample application. Callbacks to the application are used as needed. The SIP service takes care of UDP/IP socket communication with the SIP server.
In the case of the VoIP Engine Service, it too runs its own threads under control of the sample application. The VoIP Engine Service interfaces directly with the audio drivers as well as the UDP/IP stack. The host application does not need to deal with audio buffers or RTP packets.

Peer to Peer operation opens up the door to many new applications, limited only by your imagination. Download our sample app on to two or more phones and try it out for yourself.

Adaptive Digital (**easy integration + field proven algorithms**) = **quick-to-market applications**