

- + Adding PA functions to existing VOIP telephone systems
- + Adding single button call devices to a VoIP system
- + Cashier's desk in retail outlets
- + SIP based door phone panels and talk stations
- + SIP addressable loudspeaker
- + SIP based alarm systems on highways, tunnels or railway lines



## SIP Client Application

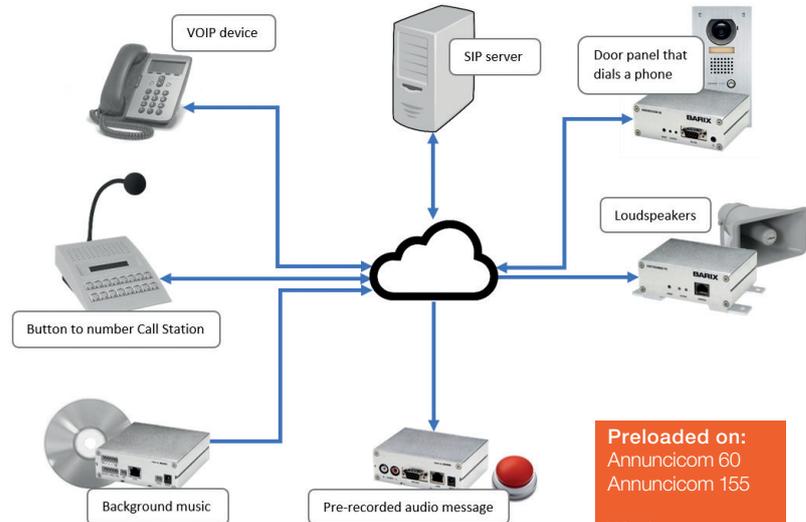
SIP Client Application is the right software to integrate Barix devices into an existing VOIP phone system. SIP Client works with most SIP based VoIP solutions including Cisco Unified Communications Manager or Asteriks and others. Using the Session Initiation Protocol (SIP), the client device receives a phone number in addition to its IP address, and can communicate either on a peer-to-peer connection or over a proxy-based connection (PBX).

Ringtone and busy tone guarantees that the call remains confidential and the recipient really receives the information. Featuring background music, audio rebroadcasting and special DTMF commands to quickly manage the device, the SIP Client is easy to integrate into an existing telephone system. Four different relay modes can be configured to open doors, windows, or to activate machines while simultaneously speaking.

A screenshot of the SIP Client web interface. The interface has a navigation bar with tabs: HOME, PROFILES, CONFIGURATION, STATUS, and DEFAULTS. The main title is "SIP CLIENT". Below the title, there are two tabs: "Basic Settings" (selected) and "Advanced Settings". There are "Apply" and "Cancel" buttons. The configuration is divided into three sections: "BASIC SETTINGS", "SIP PROTOCOL SETTINGS", and "OUTBOUND CALL SETTINGS".  
**BASIC SETTINGS**  
SIP PROTOCOL SETTINGS  
Peer to Peer:  No  Yes  
SIP Server (PBX):   
SIP ID (username):   
SIP Password (secret):   
**OUTBOUND CALL SETTINGS**  
Call on Device Inputs  
Input 0 Call ID:   
**INBOUND CALLS**  
Phone pickup mode:   
Pick/hang up time:  seconds

## FEATURES:

- + full-duplex phone communication
- + SIP (RFC 3261) compliant architecture
- + Supporting profiles for easy configuration
- + Configurable destination number to call for every input contact
- + Configurable call “pick up”/“hang off” timeout interval
- + Configurable “call/close on level” feature
- + Support for G.711 audio
- + Serial, UDP, TCP or CGI control interface
- + Priority-based notification audio messaging
- + Configurable relay to switch on at call answer/call ring
- + Transparent bidirectional Serial-To-TCP gateway
- + Friendly profile based WEB configuration UI
- + automatic network configuration (BOOTP, DHCP, IPzator)
- + DTMF door open key sequence
- + Barix X8 product support to add 8 additional inputs for predefined calls
- + Features priority notification port



**Preloaded on:**  
Annunicom 60  
Annunicom 155

**Works on:**  
Annunicom PS1  
Annunicom PS16  
Annunicom 100  
Annunicom 200  
Annunicom 1000  
Exstreamer 1xx  
Exstreamer 2xx  
Exstreamer 500  
Exstreamer P5