



Single voice channel with G.711 and G.729AB codecs  
 Conferencing handled by the gateway  
 Additional channels or codecs possible via customization with ADT  
 RTP packetization with configurable periods of 10, 20, and 30 msec.  
 Handset support with an Acoustic Echo Suppressor (AES)  
 Group listen speaker mode with volume control (Not a full duplex speaker phone)  
 RFC 2833 DTMF relay support  
 Configurable 100 millisecond jitter buffer.  
 Tone generation – DTMF or fixed tone for echoing key presses on keypad.  
 Tone generation  
 Allows for any configurable sequence of tones to be generated towards the phone or network or both.  
 Packet loss concealment compliant with G.711 appendix I  
 Comfort noise generation  
 Voice Activity Detection  
 Acoustic gain control.  
 Support for a single Ethernet interface, 3x4 keypad, hook-switch and other control switches, LEDs, and a 2x24 or similar LCD.  
 Call control & management  
 Supports RFC 3261 and 3550 SIP signaling and RTP media stream protocols.  
 Supports SIP authentication, invite, ring, ack, ok, bye and termination messages.  
 Supports all supplementary services implemented in the network / gateway  
 Supports configurable unique SIP username and password (credentials) in non-volatile memory.  
 Supports MD5 digest authentication.  
 Supports SIP user agent interoperability with proxy servers.  
 outgoing & incoming calls, call waiting, conferencing, call forward and call transfer.  
 Supports configurable digit map for dialing support.  
 Networking  
 Supports DHCP and DNS clients, IPv4/UDP/TCP.  
 Supports STUN for NAT traversal.

Features can be added to this baseline design by adding additional hardware and/or software to increase the level of functionality of the IP phone depending on the market directions and conditions.

<sup>1</sup>PLC - Packet loss concealment  
DTX - Discontinuous transmission

