

SIPivr 6800 VOIP SIP Application Platform

TIME TO HAVE VOIP VALUE ADDED SERVICE NOW!



Finally there's a SIP-based VoIP Interactive Voice Processing (IVR) developing system that help to create profitable value added service quickly and easily: **SIPivr 6800**. With easy web management interface, SIP VoIP voice resource, drag and drop call flow editor, rich set of IVR required components, real time debugger... etc., **SIPivr 6800** provides service intelligent into your VoIP network.

By using **SIPivr 6800**, the VoIP value added services such as announcement service, auto attendant, voice mail, coloring ring back tone, central IVR service, VoIP call center ... etc. can be easily adapt to your business or service platform. It creates differential service for your customers to keep your profits growing.

SIPivr 6800 are dedicated for IP based IVR applications that target to system integration company, ISR for value added service and VAR (value added reseller). Welltech also provides total SIP solution to our customers. From small voice gateway with FXS and FXO interface, SIP IP phone to E1/T1 Trunk Gateway and SIP Proxy server, are popular in the market now. This fact ensures for interoperability of all SIP devices in customer's network.

Key Features

- Up to 120 Universal VOIP Channels
- Fully Web Call Flow Editor & Debugger
- Rich-set of IVR Components
- SIP 3261 Compliance
- Support SIP Proxy Sever
- Support Call Transfer
- Voice Codec: G.723, G.729A, G.711
- Support DTMF Relay
- Support Database Connection Pools
- Support Channel Bridge
- Support Job Push & Retrieve
- Free Text Expression
- Hitless Call Flow Update
- Optimized Developing Platform
- Real Time Call Flow Debugger

Technical Specification

LAN Interface

- Two 10/100MB Ethernet ports (Host & RTP)

SIP Protocol Support

- SIP RFC 3261 compliance
- SIP RFC 3264 (Offer/ Answer)
- SIP Call on Hold
- SIP Transfer (unattended)
- SIP Transfer (attend)
- SIP/UDP
- SIP-180/SDP
- SIP-183/SDP
- SIP-PRACK

SIP Proxy Server Support

- SIP Outbound Proxy
- SIP Registrar Server

Audio Codec Support

- G.711 a/ μ -law, G.723.1 (5.3/6.3K), G.729A

DTMF Transmission

- Transparent mode
- RFC 2833
- SIP INFO

Voice Quality & Echo Cancellation

- G.168 - 2000 (echo cancellation)
- Adaptive jitter buffer
- Support silence suppression
- Gain control
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generate)





Database Component

- Database Connection Pool
- DB Connect & Disconnect
- SQL Cursor & Fetch
- SQL Transaction Protection

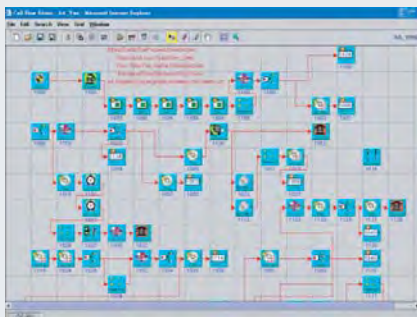
Flow Control Component

- Call Flow Start/Stop Hook Flow
- Hang-up Hook Flow
- Channel Start/Stop Hook Flow
- Sub Function
- Sub Flow
- External/Internal Hook Server Support
- Condition Case
- Go To
- Uninterrupted Protection

Basic Component

- Customizable Log
- DES & Triple DES
- Expression IF
- File & INI operation
- Free Text Mathematical Expression
- Free Text String Expression
- Job Operation
- MD5
- RADIUS AAA
- Sleep
- Time Related Component
- Variable Define
- Working Hour Decision

Call Flow Editor



IVR Component

- Answer & Hang-up
- Channel Bridge
- One channel Bridge
- Call Transfer
- Digit Manipulation
- Digit Timeout Adjustment
- Outbound Call
- Play Announcement
- Play Announcement & Collect DTMF
- Play List Support
- Send DTMF
- Voice Record
- Volume & Speed Adjustment

With SIPivr 6800,
 You are able to implement various value
 added services that expand to meet the needs
 of your business.



Call Flow Design Tool

- Full Web Management GUI
- Drag & Drop Call Flow Editor
- Real Time Call Flow Debugger
 - Breakpoint
 - Variable Watch/Edit
 - Component Trace
 - Step Debug
- Real Time Status Monitor
- Web-based Voice File Management
- Project and Channel Management

Maintenance

- Administrative Log
- Front Panel LCD Setup
- FTP/ HTTP Server
- HTTP SSL Support
- RS-232
- TELNET
- Multiple Configuration
- NTP Time Synchronization (SNTP V4)
- Password Security
- System Event Log
- Time-zone Support
- User Account Manager

Network Management

- SNMP Trap Support
- Support DNS and Dynamic DNS
- Ping
- TOS Field Setting (RTP only)
- SNMP V2 MIB I&II
- SNMP Get & Set Command
- SysLog Support

AAA

- Configurable Redundant RADIUS Server
- Provides CDR

System Limitation

- Max call flow component: 500
- Max CDR keep days: 10
- Max voice file storage: 40GB

Front Panel Display

- LED status: Power/ DOM/ System
- Front panel LCD (2 lines X 16) status display

Environmental

- 90~240V auto switch
- Operation temp: 0°C ~ 50°C
- Relative humidity: 5% ~ 95%

Physical Dimension

- Dimension: 483mm(L) x 450mm(W) x 44mm(H)
- 1U Rack mount
- Color: Black
- Weight: 7.455 Kg

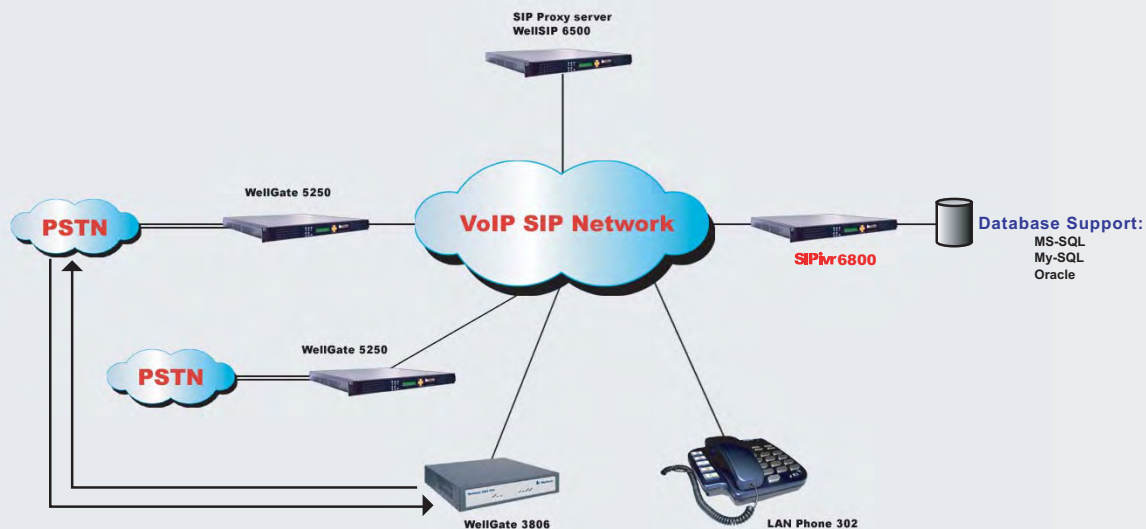


With SIPivr 6800,
You are able to implement various value added services that expand to meet the needs of your business.



00001010101000101010001010111010101

Architecture Diagram



Application

SIPivr 6800 is a platform for IP based IVR implementation, giving you the flexibility to create your own custom systems. It can be implemented a wide range of applications such as IP based IVR, pre-paid service, call back system... etc.

Order Information:

Model Number	Total of Channels
SIPivr 6800 - 30	30 Universal VoIP IVR Channels
SIPivr 6800 - 120	120 Universal VoIP IVR Channels

