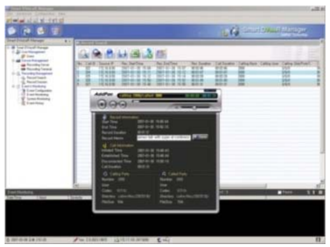
IPNext600
Call Center Software Features

IPNext600 SIP Call Manager

Software Features for
Call Center Service



AP-NR1500
IP Voice Recording Server

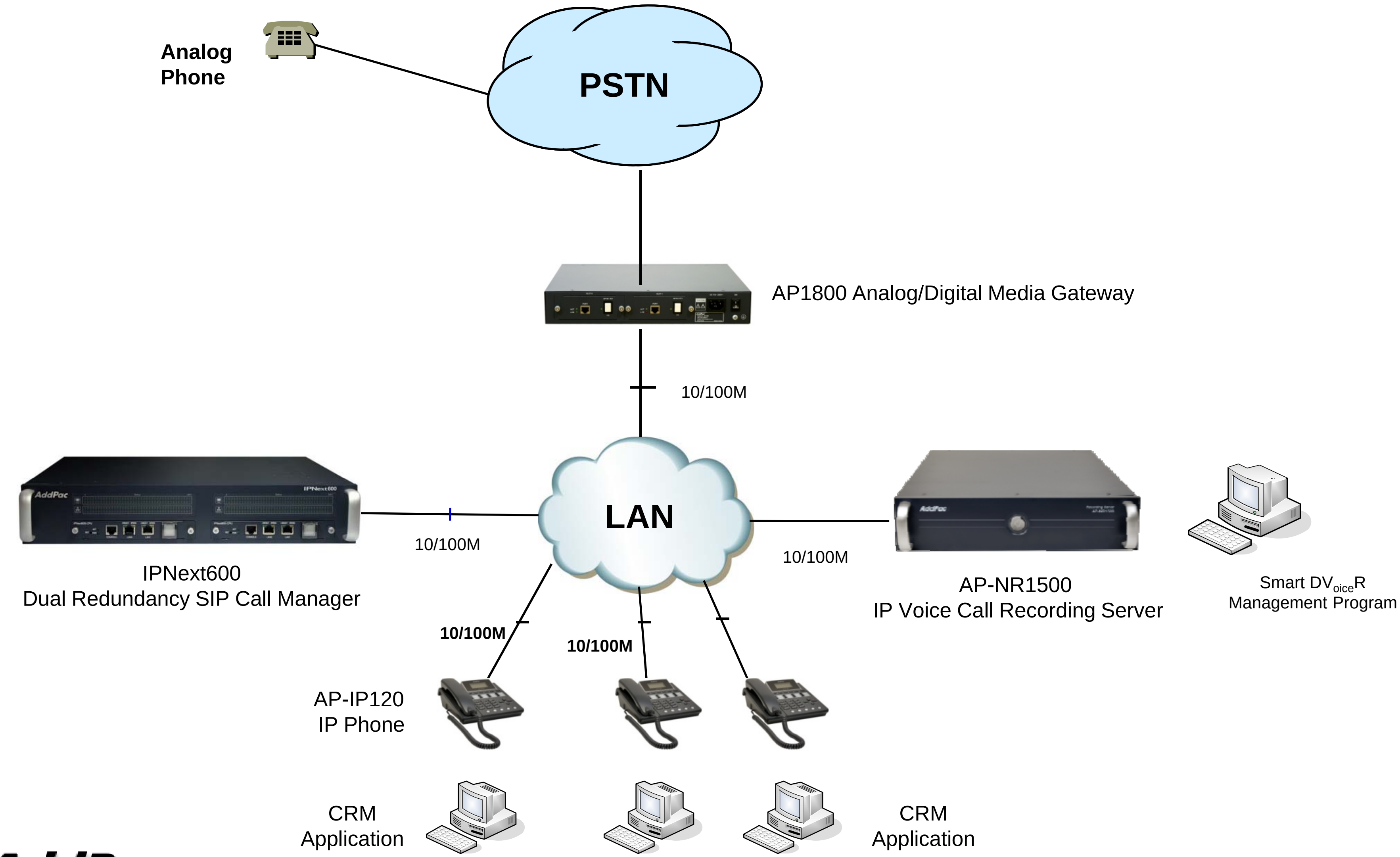
AddPac Technology

Sales and Marketing

www.addpac.com

Contents

Network Diagram



IPNext600
SIP Call Manager

Main Features

- SIP Application Server, Proxy, Registrar and Location Server
- Multiple ITSP Trunk with SIP & H.323 Accounts Support
- Dual System Redundancy Architecture
  - Two(2) Fast Ethernet Interface / System
- High Performance RISC Architecture
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- IPv4/IPv6 Dual Stack
- RTP Proxy Function Embedded for Private IP and IPv6 Address Interworking
- User Presence Service Features for Smart Multimedia Messenger and Smart IP Phone
- IVR Scenario Editor, Voice Mail, Media Service (Coloring), Conference
- Firmware Upgradeable Architecture
- Smart Multimedia Manager for IP-PBX Management
- Smart Messenger Service (click to dial) for Unified Communication
- VPMS (VoIP Plug&Play Management System) & Smart NMS for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Dual Redundancy Power Module

Hardware Specification

RISC
CPU

- High-End Microprocessor Computing Power
- Main Chassis
  - Dual Redundancy CPU Boards for System Fault Tolerant
    - Two(2) 10/100Mbps Gigabit Ethernet
    - One(1) RS-232C Console (RJ45)
  - Dual Redundancy Power Supply Module
  - Hot-Swap Features

Hardware Specification

IPNext600 Call Manager

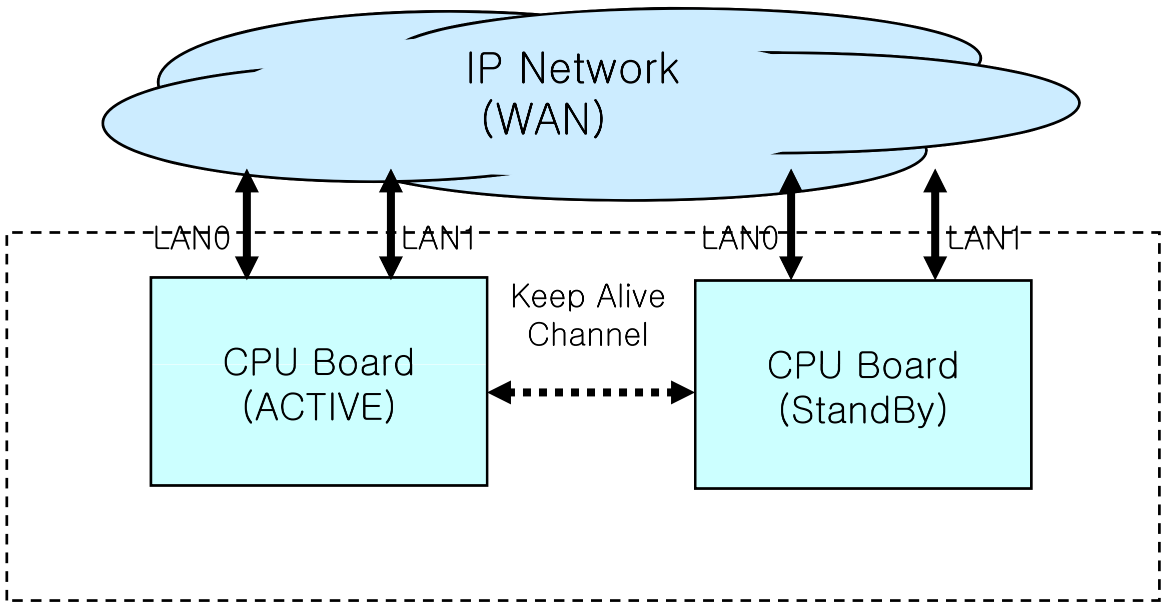
RISC
CPU

Front Side

System Redundancy Features

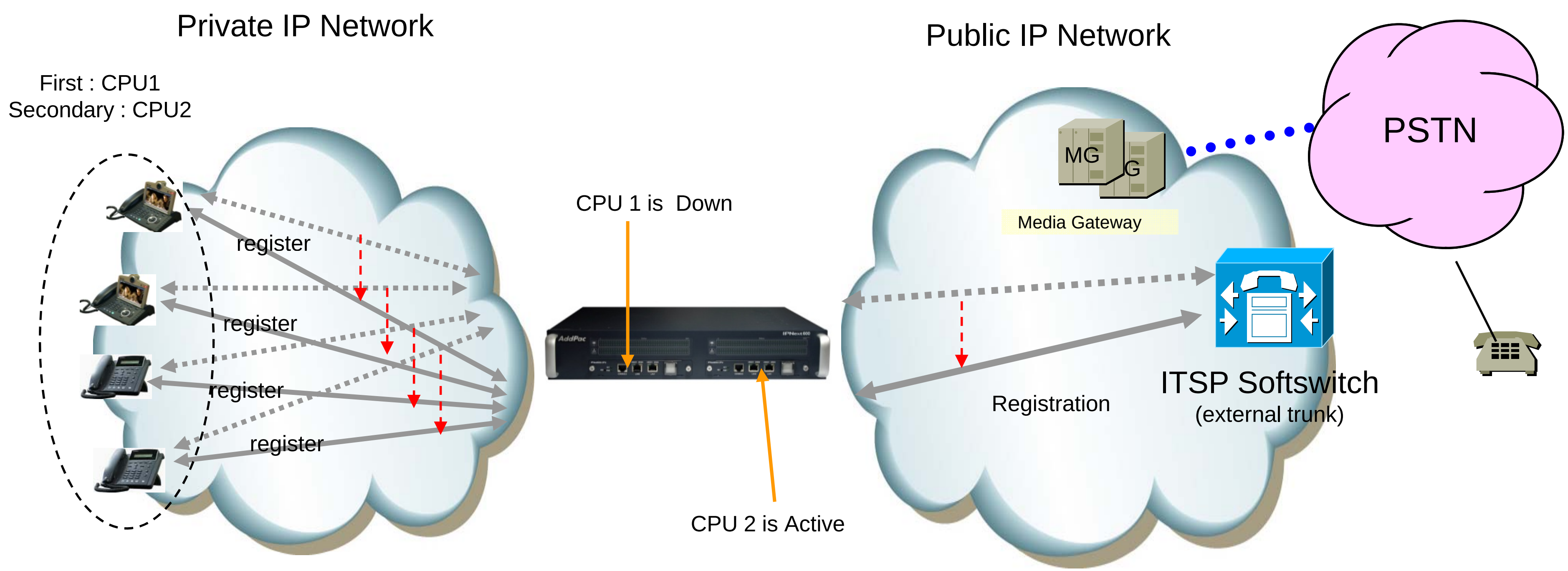
IPNext600 Call Manager

IPNext600 System Block Diagram



System Redundancy Features

- Active– Active Duplication Scheme
- Active – Standby Duplication Scheme
- VRRP based Duplication Scheme



Active – Standby Duplication Scheme (example)

IP Telephony Service and Features

- **Signaling Server**
  - SIP Application Server, Proxy, Registrar and Location Server (RFC3261)
  - Multiple ITSP Trunk with SIP & H.323 Accounts Support
    - **IP UA Client Role for Registering to ITSP SIP Server**
    - **H.323 Gatekeeper Client Role for Registering to ITSP H.323 Gatekeeper Server**
- **IVR & Auto Attendant**
  - Default Auto Attendant Support
  - Interactive Voice Response (IVR)
    - **Provides with GUI-based Smart IVR Scenario Editor**
    - **Upload/Download Scenario by Smart IVR Scenario Editor**
    - **Supports Multiple Concurrent Scenarios**
    - **Supports Recordable IVR Prompts**
- **Voice Mail**
  - Support Voice Mail with IVR
  - Access from Remote Site via Trunk Support
  - Voice Mail Notification Support

IP Telephony Service and Features

- **Number & Call Routing**
  - Trunk Hunting by Preference or Sequential
  - Call Hunting by Preference, Simultaneous, Random
  - Call Hunting by Chained Hunting Group
  - Partition for Address Grading
  - Call Class for Call Access Control
  - Number Translation Rule for Inbound/Outbound Call
  - Centrex with Prefix Support
  - Multiple Shared Devices with One Number
  - Multiple Numbers on One Device
  - Individual Call Park within Park Number Pool
  - Group Call Park within a Group or Other Group
  - Call Pickup of Ringing Call of Same Group or Other Group
  - Call Pickup of Parked Call
  - Call Transfer - Blind, Consult
  - Call Forwarding - Unconditional, Busy, No Answer, Voice Mail
  - Call Waiting
  - Call Swaping
  - Call Hold

IP Telephony Service and Features

- **IP-PBX Advanced Features with AddPac IP Phones**
  - Multiple Call Handling with Call Status and Calling Line Number and Name
  - Plug and Play with Auto Discovery Function
  - Softkey Map Download and Control
  - Time and Date Setting
  - Voice Mail List View
  - Parked Call List View
  - Call Forward Setting
  - Recent Call List View
  - Calling Number and Name Identification
  - Individual Call Park within Park Number Pool by Softkey
  - Group Call Park within a Group or Other Group by Softkey
  - Call Pickup of Ringing Call of Same Group or Other Group by Softkey
  - Call Pickup of Parked Call by Softkey
  - Call Transfer - Blind, Consult by Softkey
  - Call Waiting Indication
  - Call Swaping by Softkey
  - Call Hold by SoftKey
  - Conference Control

AP1800
Analog/Digital Media Gateway

Product Overview

- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- Analog/Digital VoIP Gateway Solution
- Various Analog Interface Support : FXS, FXO, E&M
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Performance LAN-to-LAN Routing Capability
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- Smart Network Management for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with Internal Power Supply
- Two(2) VoIP Module Slot
- Hot-Swap Function Support

Hardware Specification

RISC
CPU

High-end
DSP

- **RISC Microprocessor Computing Power**
- **Up to 16 Port Analog VoIP Gateway**
- **Two(2) VoIP Module Slots (Hot-Swap)**

  - 8-Port FXS Card, 8-Port FXO Card, 4-Port FXS 4-Port FXO Card , Digital E1/T1 Card
- **Network Interface**

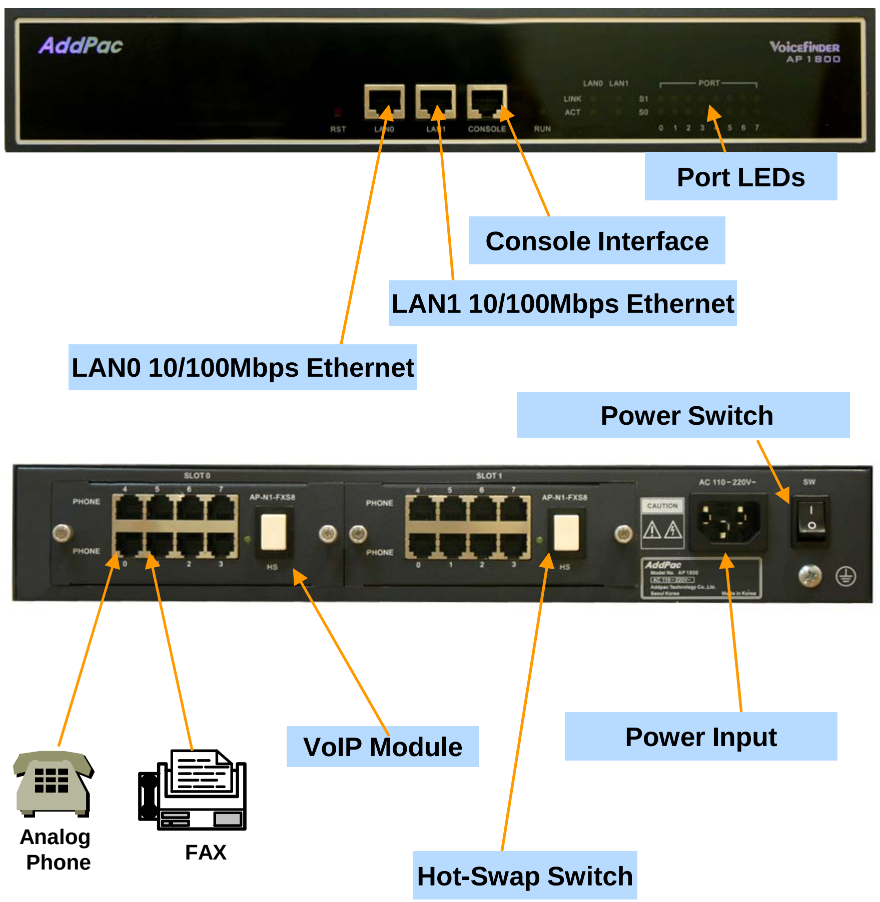
  -Two(2) 10/100Mbps Fast Ethernet (RJ45)
- **RS232C Console Interface**
- **Run LED, LAN LED, Port LEDs**
- **Internal Power Supply**

Hardware Specification

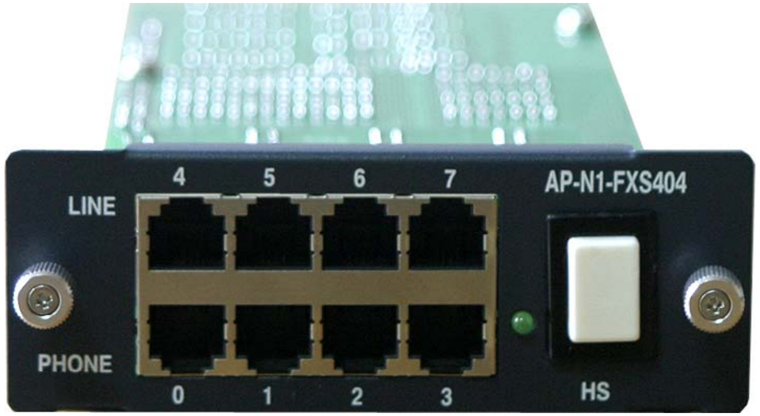
Hardware Specifications

AP1800 VoIP Series	Basic Specifications
Voice Interface	Two(2) VoIP Module Slots
	AP-N1-FXS8, AP-N1-FXO8, AP-N1-FXS4O4, AP-N1-E1
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	32 Mbyte High-speed SDRAM
Power Requirement	Power Supply Adaptor / VAC 110~220V, 50/60Hz,
Operating Temperature	0ºC ~ 45ºC (32 ºF ~ 122ºF)
Storage Temperature	-40ºC ~ 85ºC (-40ºC ~ 185ºF)
Relative Humidity	5% ~ 95% (Non-condensing)

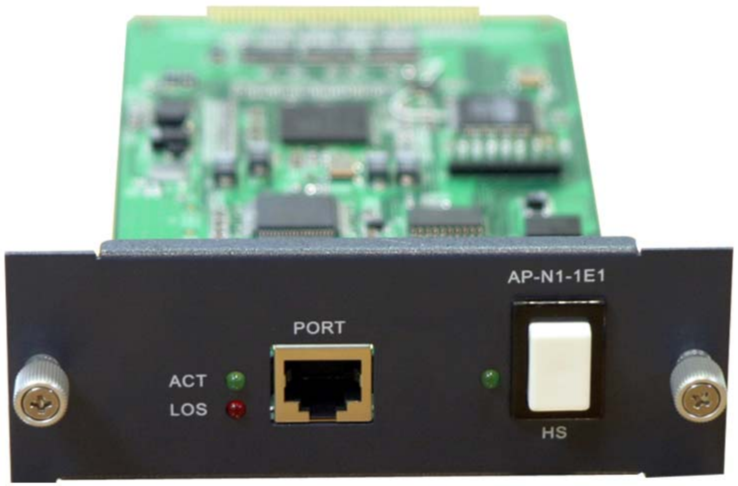
Network interface Configurations



Hardware Specification



AP-N1-FXS4O4
(4-Port FXS-
4-Port FXO Module)



AP-N1-E1
(Digital E1 Module)

AP-IP120 IP Phone

Product Overview

- IP Phone Solution
- 12 Speed-Dial Key with Presence Indication Lamp
- Audio Broadcasting Solution
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

Hardware Specification

- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- Optional PSTN Backup (FXO) Interface
- Optional PoE (Power over Ethernet)
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio Output Connector
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet
- LCD Window : Graphic LCD (4 Line Text)
- 12 Speed-Dial Key with Presence Indication LAMP
- Power Supply
  - External Power Adaptor (5V, 2A)

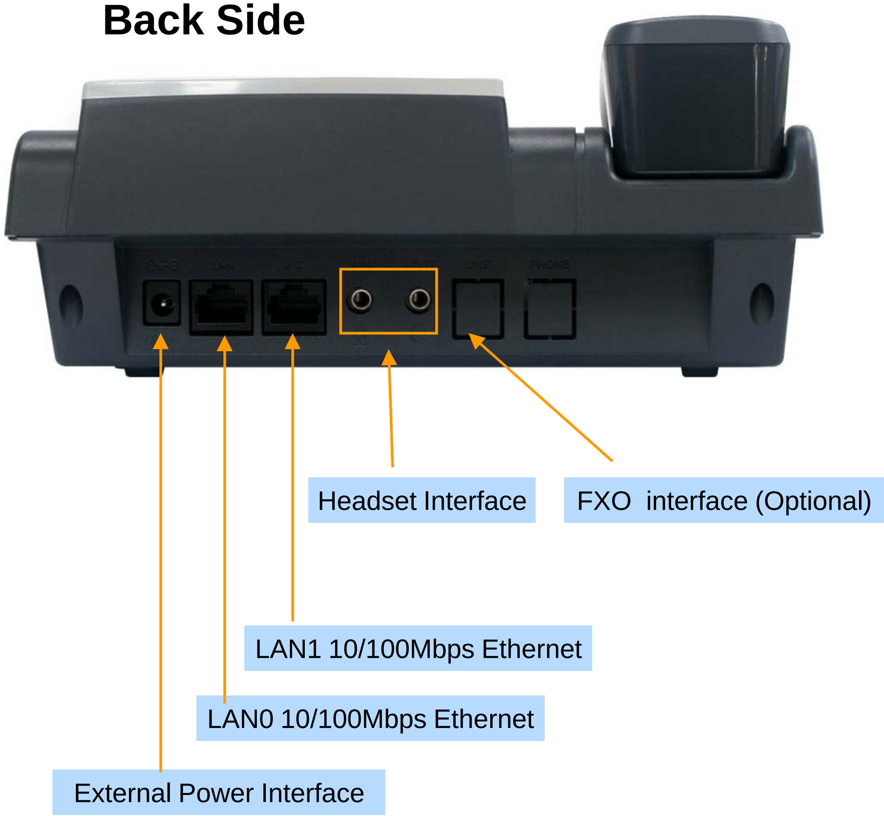
Hardware Specification

Hardware Specifications

AP-IP120 IP Phone	Basic Specifications
CPU	RISC Microprocessor
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
PSTN Backup Port (Optional)	1-Port PSTN Backup Port(RJ-11)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	16Mbyte High-speed SDRAM
Power Requirement	External Power Supply Adaptor / VAC 110~220V, 50/60Hz, 10Watt(5V,2A)
	Power over Ethernet (option)
Operating Temperature	0ºC ~ 45ºC (32 ºF ~ 122ºF)
Storage Temperature	-40ºC ~ 85ºC (-40ºC ~ 185ºF)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimensions	H x W x D (70mm x 200mm x 210mm)
Weight (g)	1Kg

Network interface Configurations

Back Side

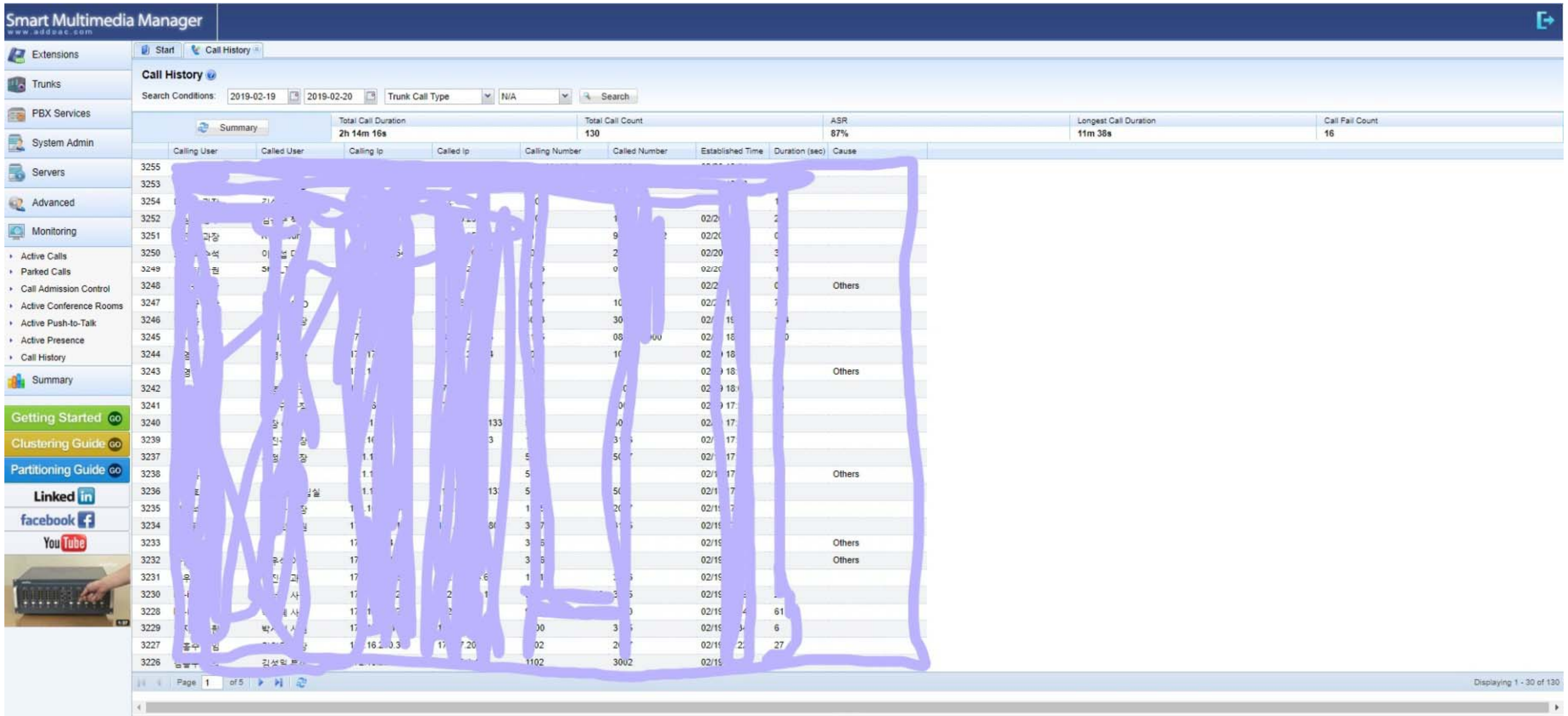


Software Features for Call Center Service

Contents

- Call Log
- IVR Scenario Editor
- CRM API
- Call Hunt Group (Enough for Small Call Center)
- ACD

Call Log (Main)

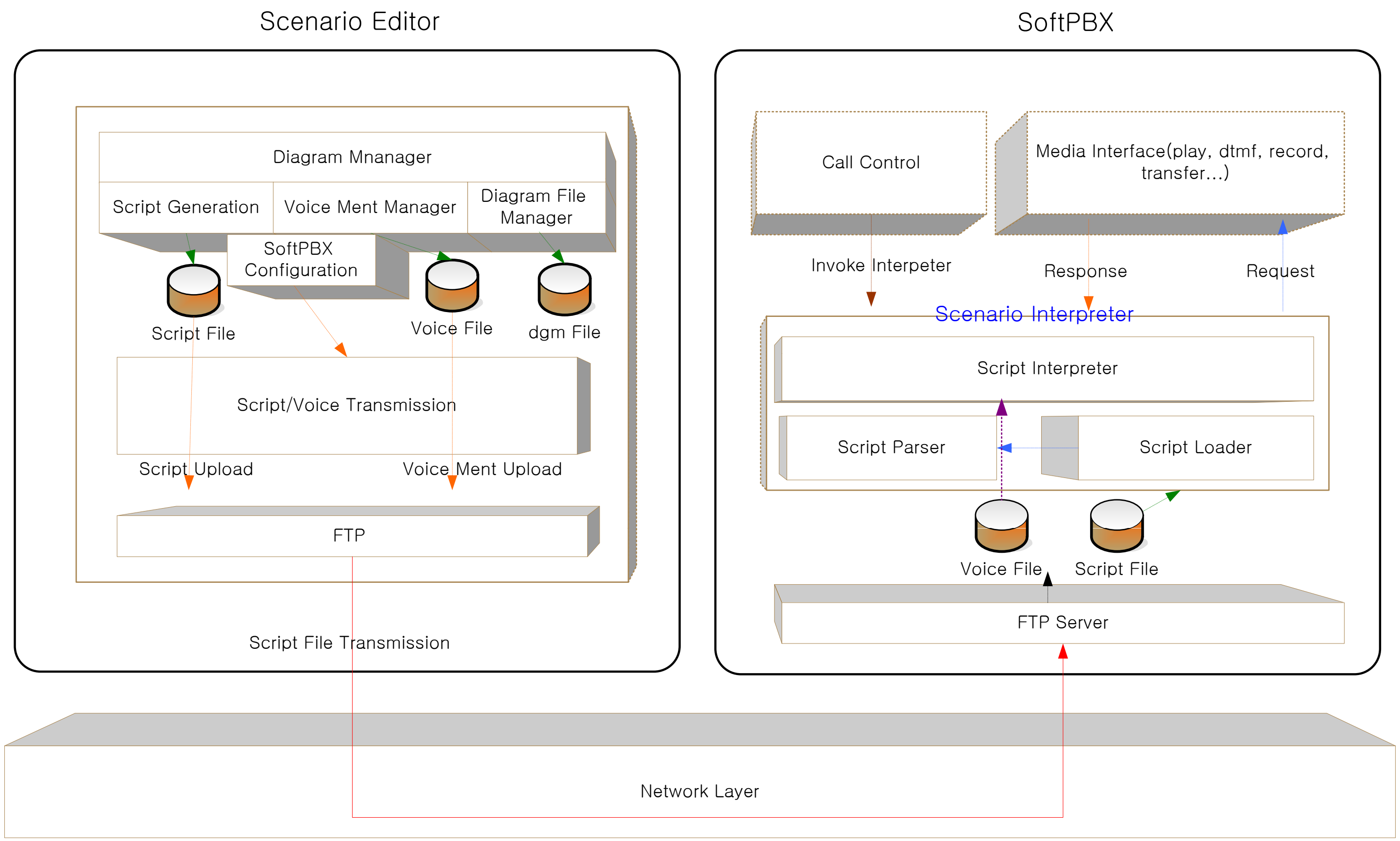


Call Log (Search Condition)

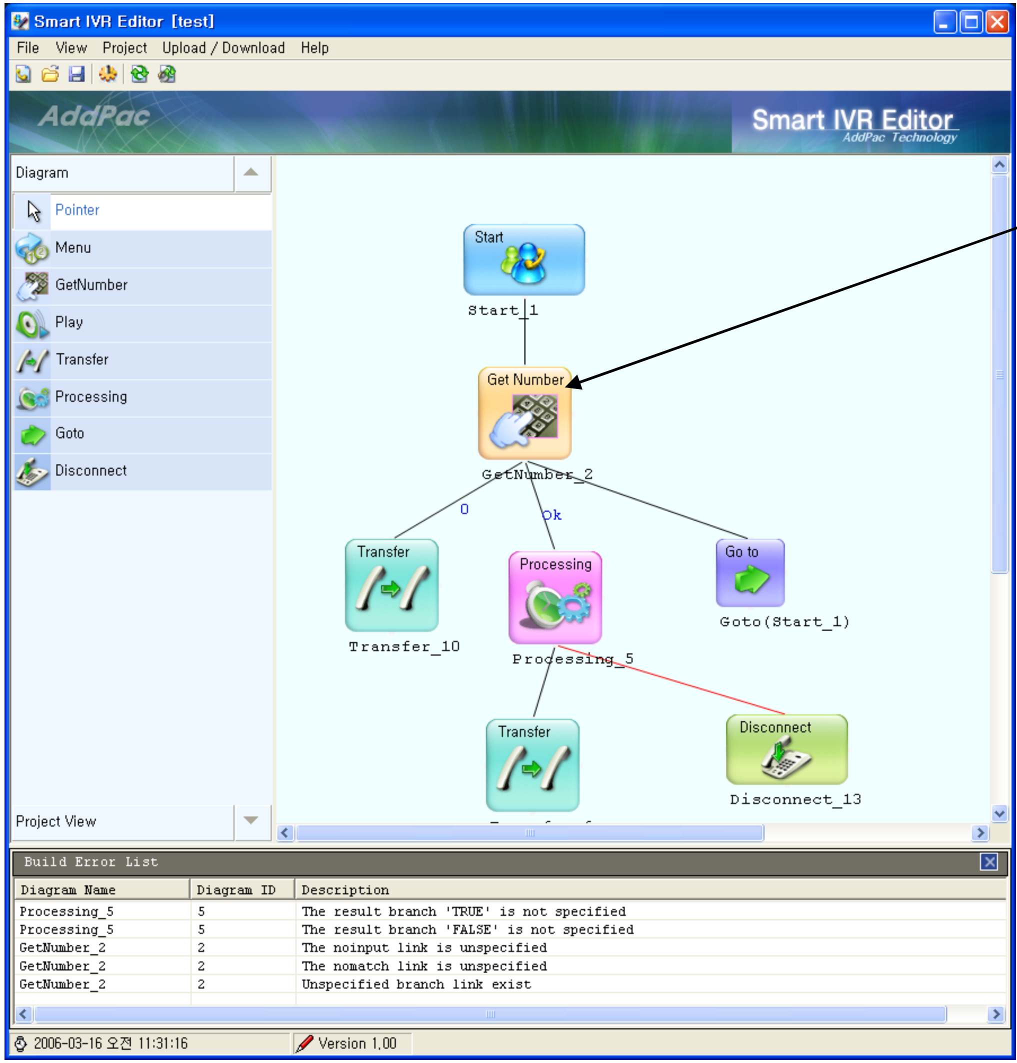
- Search Condition
  - Date
  - Trunk Call Type
    - NA
    - Unspecified
    - Inter-Site Call
    - PSTN Backup
    - Service Provider
  - User Name
  - Phone Number

IVR Scenario Editor

IVR Scenario Editor Architecture



IVR Scenario Editor Creation



Component

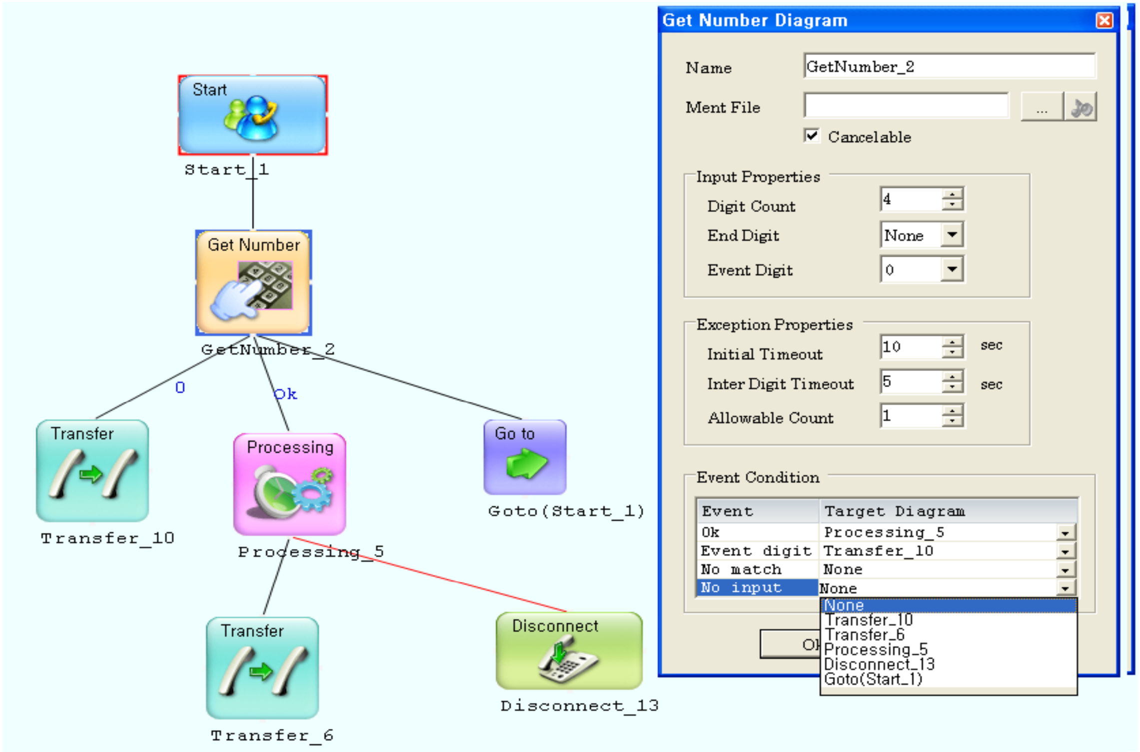
IVR Scenario Editor Creation

IPNext180 IP keyPhone System

- GUI based IVR Scenario Editor
- Support Pre-defined Component (DTMF Input, Call Transfer, Voice File Play, etc)
- Support Project Template File for Easy Modification and Reference.
- Support Pre-Defined IP-PBX System API and Additional Customization API
- Support IVR Scenario Creation Error Debugging Features

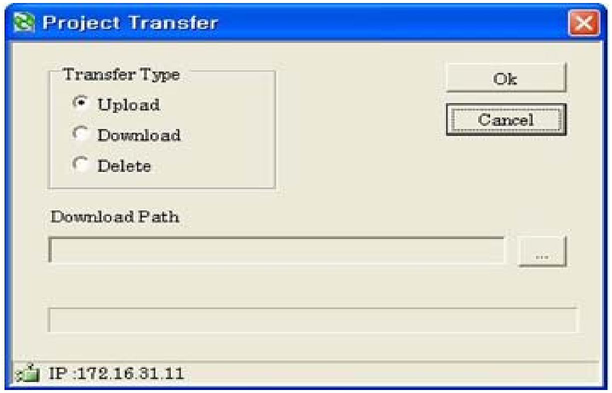
IVR Editor Component Property

IPNext180 IP keyPhone System

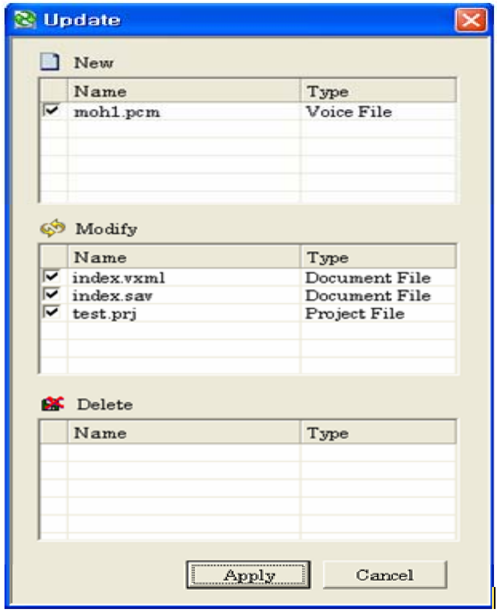


- Support the Property Setting for Component Flow Diagram (Input Event, Exception Properties, Event Condition)
- Provides the different IVR Component Flow depend on Event Condition

IVR Scenario Management



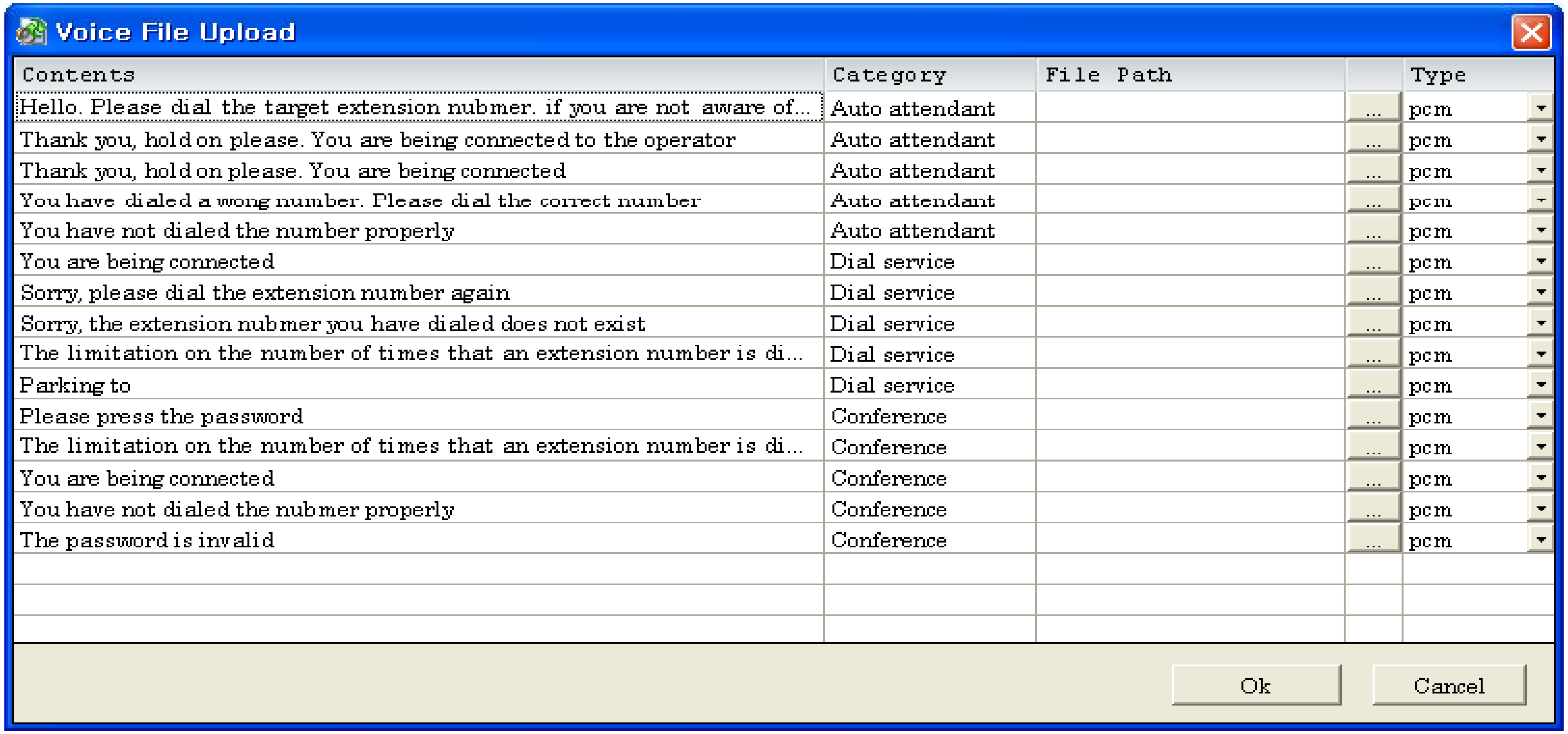
Project Transfer



Project Update

- IVR Scenario Script File can be upload or download to (from) IP-PBX.
- IVR Scenario Script File Version Control (Update, Add and Delete)
- Register Service with Smart Multimedia Manager

Voice Announcement File Management



Contents	Category	File Path		Type
Hello. Please dial the target extension nubmer. if you are not aware of...	Auto attendant		...	pcm
Thank you, hold on please. You are being connected to the operator	Auto attendant		...	pcm
Thank you, hold on please. You are being connected	Auto attendant		...	pcm
You have dialed a wong number. Please dial the correct number	Auto attendant		...	pcm
You have not dialed the number properly	Auto attendant		...	pcm
You are being connected	Dial service		...	pcm
Sorry, please dial the extension number again	Dial service		...	pcm
Sorry, the extension nubmer you have dialed does not exist	Dial service		...	pcm
The limitation on the number of times that an extension number is di...	Dial service		...	pcm
Parking to	Dial service		...	pcm
Please press the password	Conference		...	pcm
The limitation on the number of times that an extension number is di...	Conference		...	pcm
You are being connected	Conference		...	pcm
You have not dialed the nubmer properly	Conference		...	pcm
The password is invalid	Conference		...	pcm

Voice File Upload

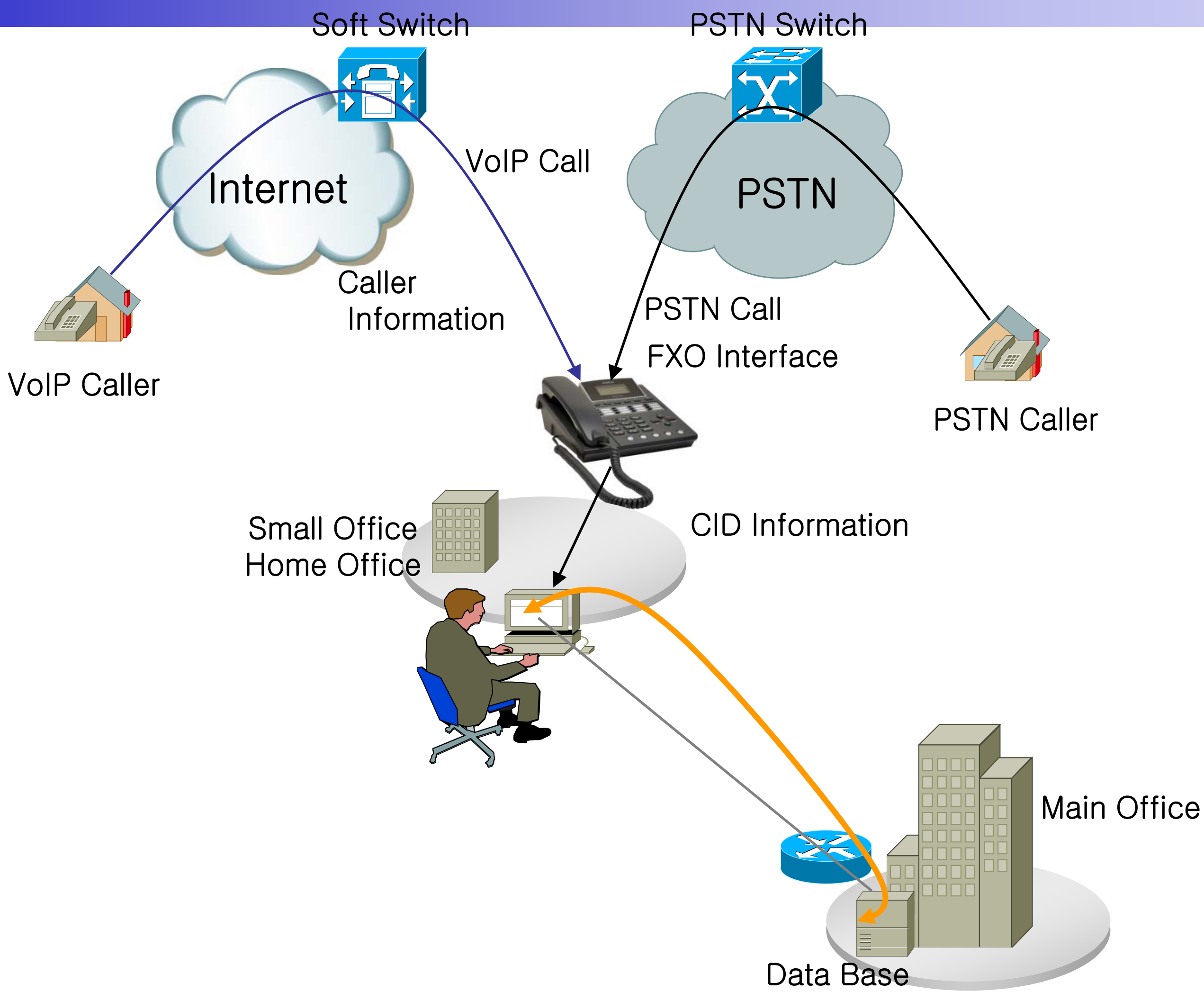
- Voice Announcement File Upload and Download for Backup

CRM API

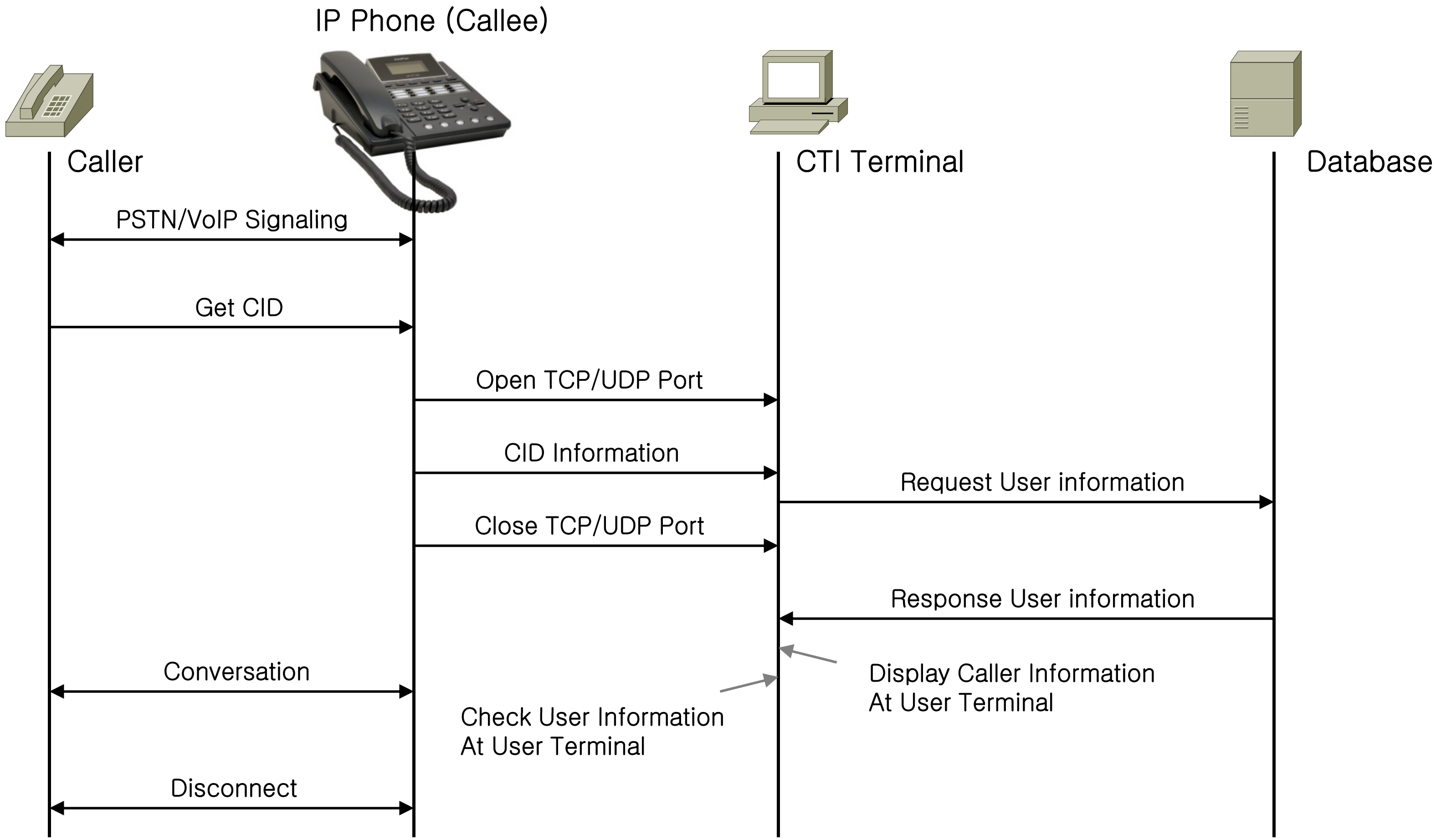
CTI using CID

- Simple CTI (Computer and Telephony Integration) Application
- AddPac IP-Phone or VoIP Gateway send CID information to CTI application via TCP/UDP socket
- CTI application get caller information using HTTP or custom specific protocol

Basic Network Diagram



Message Flow



ACD (Automatic Call Distribution) and Call Hunt Group

ACD (Automatic Call Distribution) on Attendant Queue

- The Attendant Queue is used for attendants of an organization or a call center
- When a call is inbound to the Attendant Queue, the call will be queued and distributed to one of queue member
- ACD policies
  - Longest Idle Time: Call will be distributed to longest idle queue member
  - Preference: Call will be distributed by preference order of queue member
  - Sequential: Call will be distributed to next queue member sequentially

Add an Attendant Queue Web UI

User Extension
A user extension is an IP Phone (SIP / SSCP phone) or a soft phone for end user. (The SSCP is enhanced SIP with XML based feature control protocol.)

Batch Job for User Extensions
Gives you simple and automated way to add, modify or delete one or more extensions through CSV (Comma Separated Values) file. Each CSV file can be created with your favorite text editor or Microsoft Excel.

Hunt Group
A hunt group has members of user extensions. Within a hunt group, an available member (user extension) can receive a call to the hunt group extension. A hunt group has one of simultaneous, sequential or random call hunting mode.

Pickup Group
A pickup group has members of user extensions who can pick up a ringing call within the group. The pickup group extension number is used for picking up a call by other group member.

Park Pool
A park pool is a set of extensions for parking calls. When a user parked an active call, an extension in this pool will be assigned. Other user can pick up the parked call using the parked extension number.

Conference Room
A conference room extension is used for making a conference room. The conference room can be open by WSMM or User Portal web page or by call to conference room number by privileged user (chair or operator) or by schedule. In case of dial-out participants, they receive call when conference is opening. In case of dial-in participants, they have to make a call to conference extension to join to opened conference.

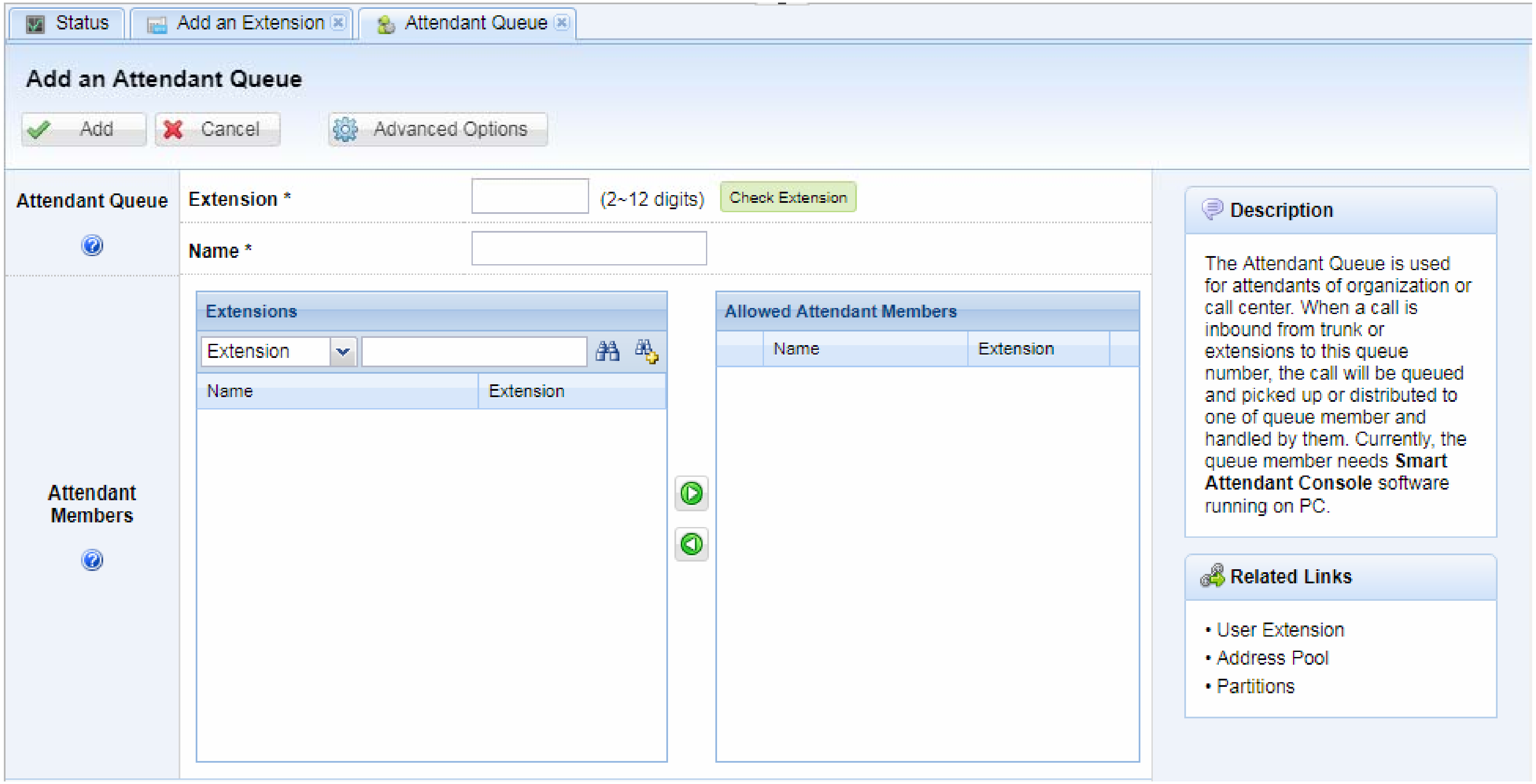
IVR Extension
An IVR (Interactive Voice Response) extension has a role of auto attendant for incoming calls from trunks. If incoming calls from trunk are routed to an IVR extension by incoming call rule, the interactive scenario will be proceed to transfer the call to a proper user extension.

Push-to-Talk Group
A PTT (Push to Talk) group has members of user extensions who will receive broadcasting announcement with auto answering and also can be a floor (speaker role) by pushing the talk button. This is half-duplex two-way broadcasting.

Paging Group
A paging group has members of user extensions who will receive broadcasting announcement with auto answering by speaker phone. This is half-duplex one-way broadcasting.

Attendant Queue
The Attendant Queue is used for attendants of organization or call center. When a call is inbound from trunk or extensions to this queue number, the call will be queued and picked up or distributed to one of queue member and handled by them. Currently, the queue member needs **Smart Attendant Console** software running on PC.

Attendant Queue Web UI



Attendant Queue Web UI

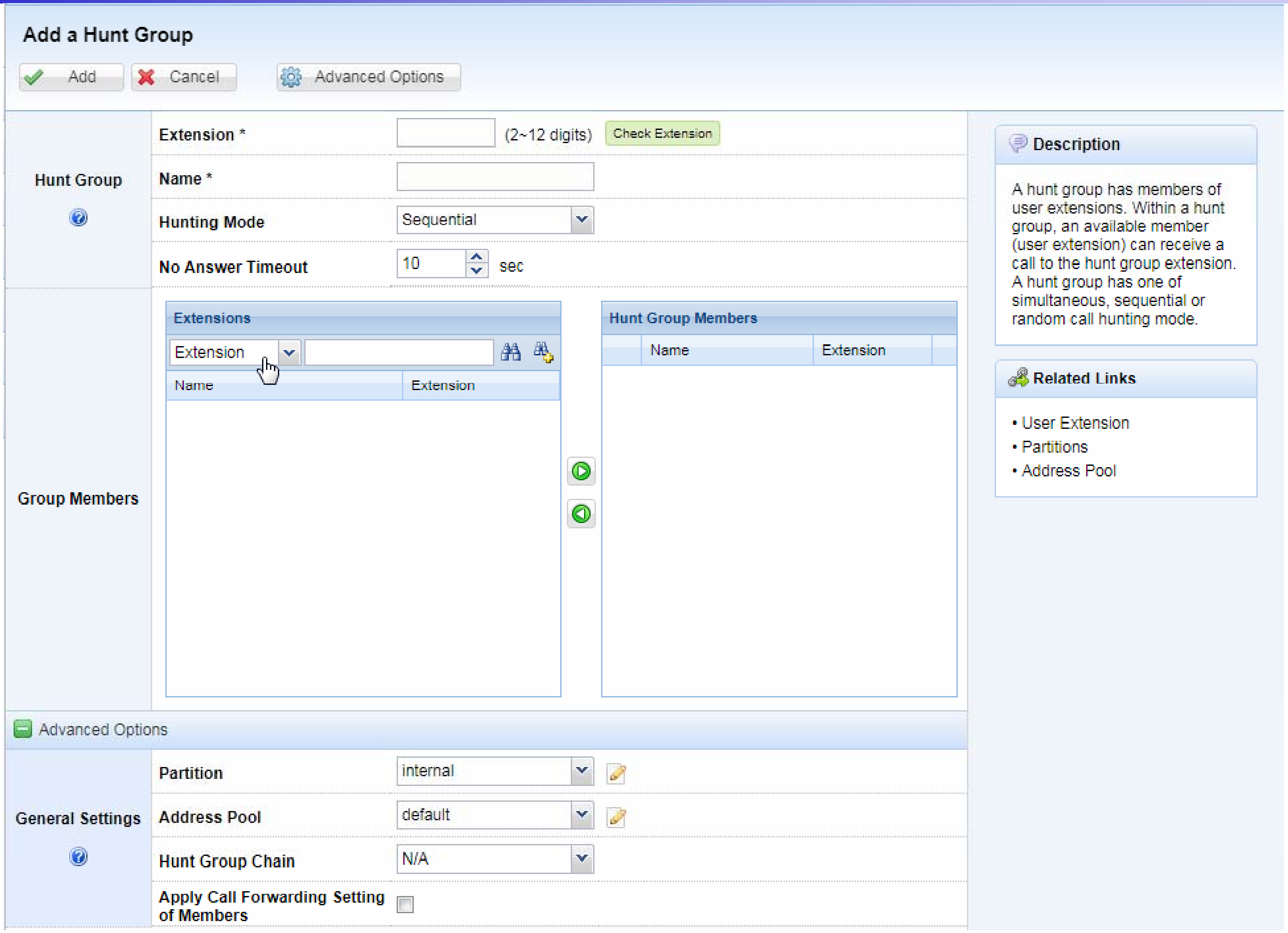
Advanced Options

General Settings	Partition	internal
	Address Pool	default
Queue Policy	Queue Size	1024
	Queue Fulled Mode	Announcement
	Redirection Target Number	
Hold Time Policy	Max Hold Time	100
	Hold Time Expired Mode	Announcement
	Redirection Target Number	
Call Distribution	Enable Call Distribution	☐
	Automatic Call Distribution	Longest Idle Time

Call Hunt Group

- A hunt group has members of user extensions. Within a hunt group, an available member can receive a call
- Call Hunting Mode
  - Preference
  - Simultaneous
  - Random
- Call Hunting by Chained Hunting Group

Hunt Group Web UI



Difference between Attendant Queue and Hunt Group

- The Attendant Queue is similar to the Hunt Group
- The Attendant Queue accepts an incoming call even if all attendant members are busy. The queued call will be distributed to a member when the member is available
- The Hunt Group rejects an incoming call when all members in the group are busy

Thank you!

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