

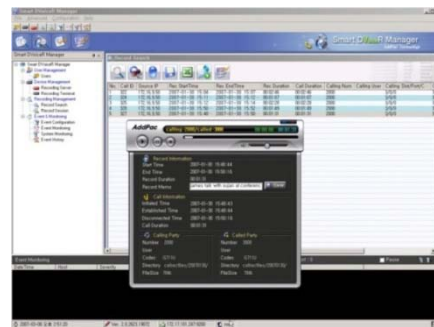
IPNext190

IP Call Center Software Features

IPNext190 Hybrid IP-PBX



AP-NR1500
IP Voice Recording Server



AddPac

AddPac Technology

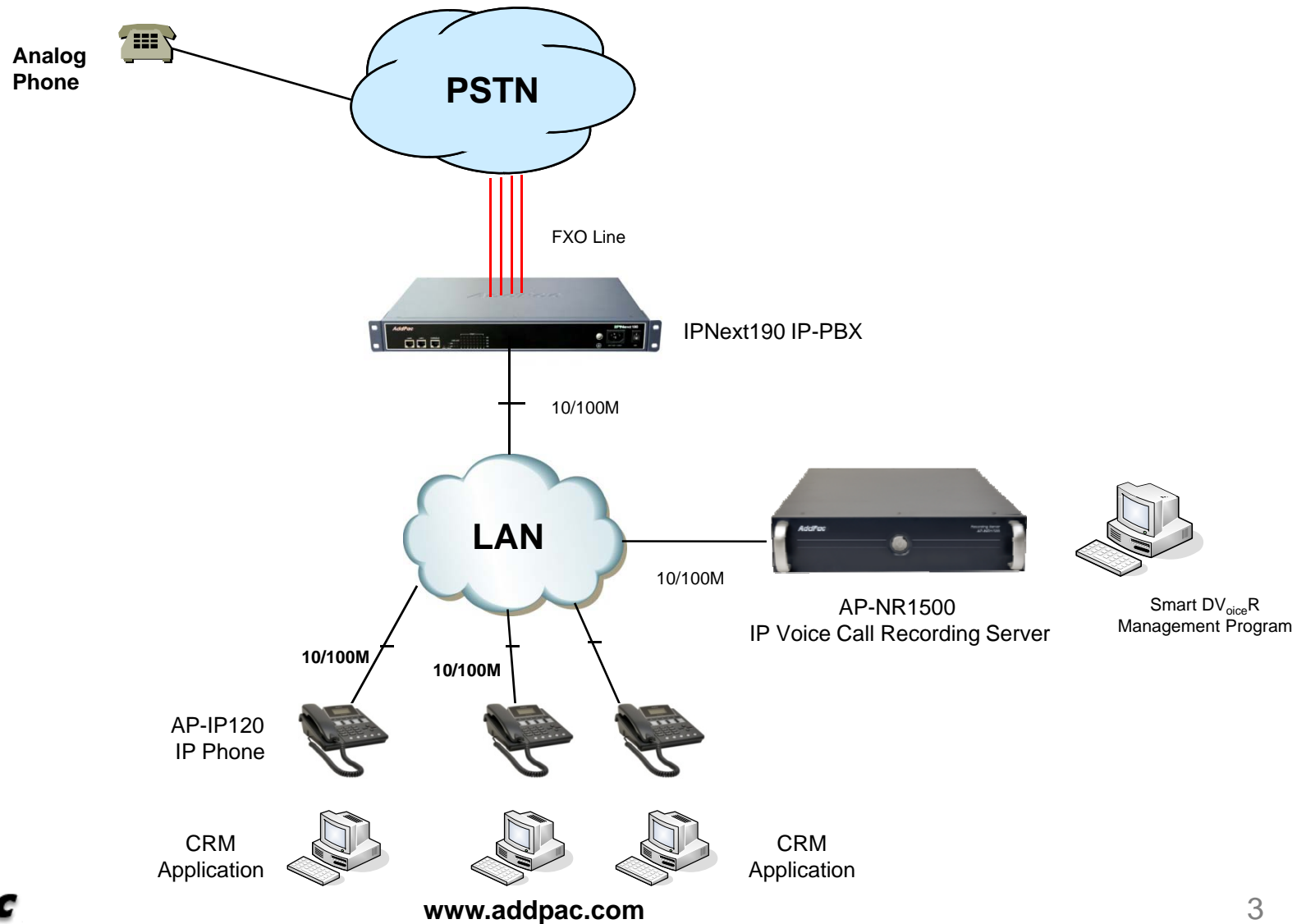
Sales and Marketing

www.addpac.com

Contents

- Network Diagram
- Small Scale IP Call Center Solution
 - IPNext180 Hybrid IP-PBX
 - AP-IP120 IP Phone
- Software Features for Call Center Service
 - Call Log
 - IVR Scenario Editor
 - CRM API
 - ACD, Hunt Group

Network Diagram



IPNext190 NGN Hybrid IP-PBX System



Product Overview

- IP based Advanced Hybrid IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Small Office
- PSTN Interface (FXO, FXS, E1/T1) Support
- IP Telephony Call Center Software Features (Call Log, ACD, IVR, Voice Recording, etc)
- Powerful Management and User Friendly Features
- Fault Tolerant and Scalability Architecture
- High-performance Video, Audio, and Voice Service
- Firmware Upgradeable Architecture
- IVR Service with Scenario Editor
- Voice Mailing Service
- Presence Service for High-end IP Phone, UC
- RTP Proxy Service for Private IP service
- SIP, H.323 Signaling for Outbound Calls
- Various Call Scenario (Call Pickup, Call Park, Call Transfer, etc)
- Various IP Terminal Support

Hardware Specification

64bit
CPU

DSP

Front Side



LAN1 (10/100Mbps)
LAN0(10/100Mbps)
Console Port

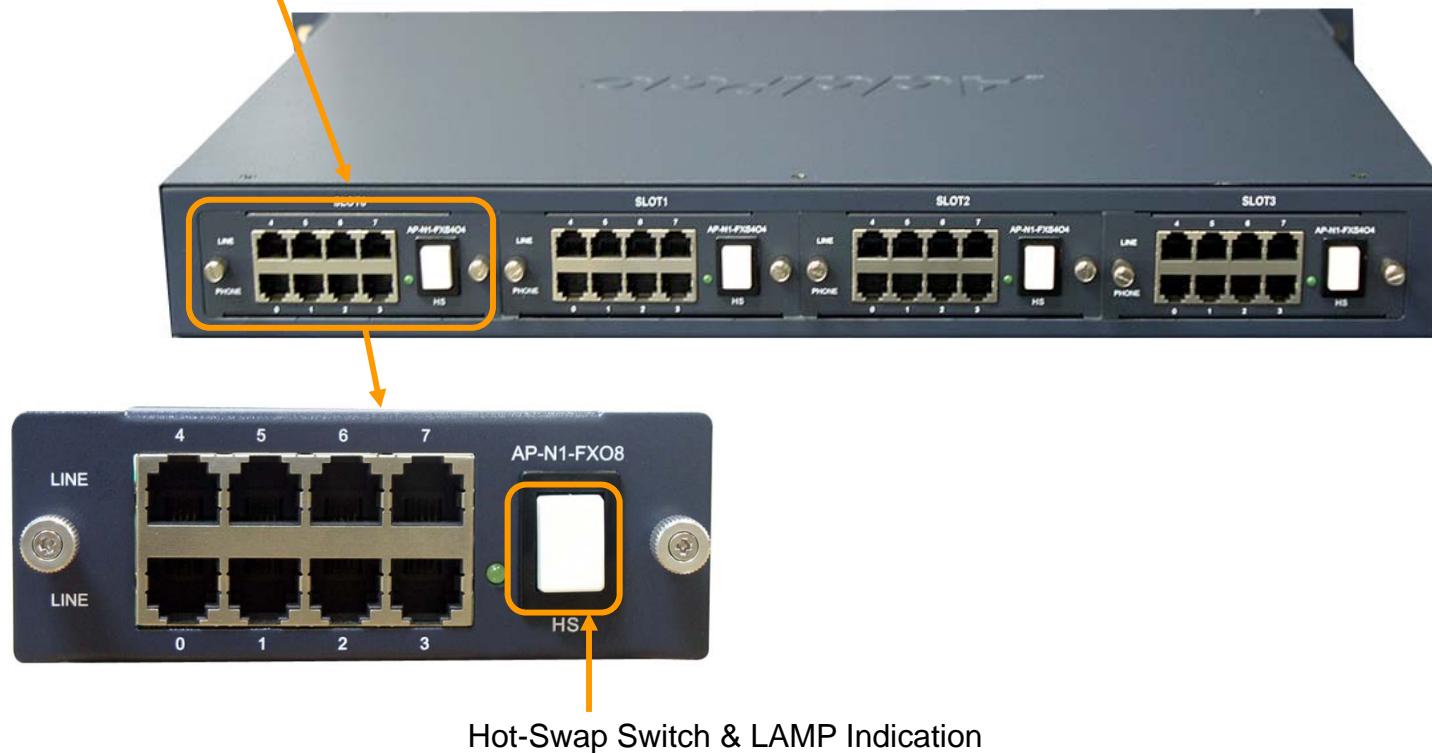
Power Inlet
Power Switch

Hardware Specification

64bit
CPU





DSP

PSTN Interface Module Back Side



Hardware Specification

- **VoIP Interface Module**

AP-N1-FXS8		8-Port FXS Voice Processing Module (8 x RJ11)
AP-N1-FXO8		8-Port FXO Voice Processing Module (8 x RJ11)
AP-N1-FXO4S4		4-Port FXO and 4-Port FXS Voice Processing Module (8 x RJ11)
AP-N1-E1T1		1-Port VoIP Digital E1/T1 Interface Module(1xRJ45)

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

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

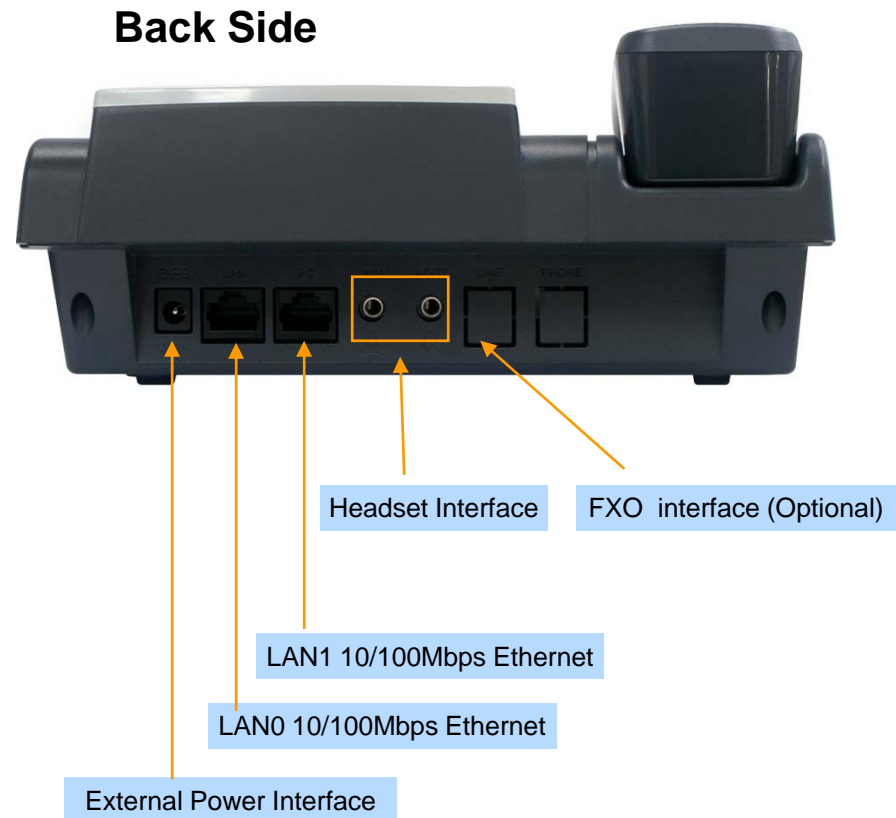
- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- Optional PSTN Backup (FXO) Interface
- Optional PoE (Power over Ethernet)
- High quality Audio and Voice Interface
 - Stereo Audio Input Connector
 - Stereo Audio Output Connector
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet
- 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	0°C ~ 45°C (32 °F ~ 122°F)
Storage Temperature	-40°C ~ 85°C (-40°C ~ 185°F)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimensions	H x W x D (70mm x 200mm x 210mm)
Weight (g)	1Kg

Network interface Configurations



Hardware Specification

- High Performance Computing Power
- Network Interface
 - One(1) 10/100/1000Mbps Gigabit Ethernet Port
- Two(2) USB 2.0 Interfaces for Mouse, Secondary Storage, etc
- One(1) RS232C Console Interface (RJ45)
- Up Two(2) SATA type Hard Disk (4~8 Tera HDD Capacity)
- Power On/Off Soft Switch with LED Indication Lamp (Front 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)

Smart Multimedia Manager
www.addpac.com

Start Call History

Call History

Search Conditions: 2019-02-19 2019-02-20 Trunk Call Type N/A Search

Summary

Calling User	Called User	Calling Ip	Called Ip	Calling Number	Called Number	Established Time	Duration (sec)	Cause
3255								
3253								
3254								
3252						02/20	2	
3251						02/20	0	
3250						02/20	2	
3249						02/20	1	
3248						02/20	0	Others
3247						02/20	1	
3246						02/18	30	
3245						02/18	08	00
3244						02/18	10	
3243						02/18		Others
3242						02/18		
3241						02/17	01	
3240						02/17	40	
3239						02/17	3	
3237						02/17	56	
3236						02/17	5	Others
3236						02/17	13	5
3235						02/17	1	
3234						02/17	06	3
3233						02/15	3	8
3232						02/15	3	8
3231						02/15	1	
3230						02/15	3	9
3228						02/15	2	61
3229						02/15	3	6
3227						02/11	2	27
3226						02/15	1102	3002

Page 1 of 5

Displaying 1 - 30 of 130

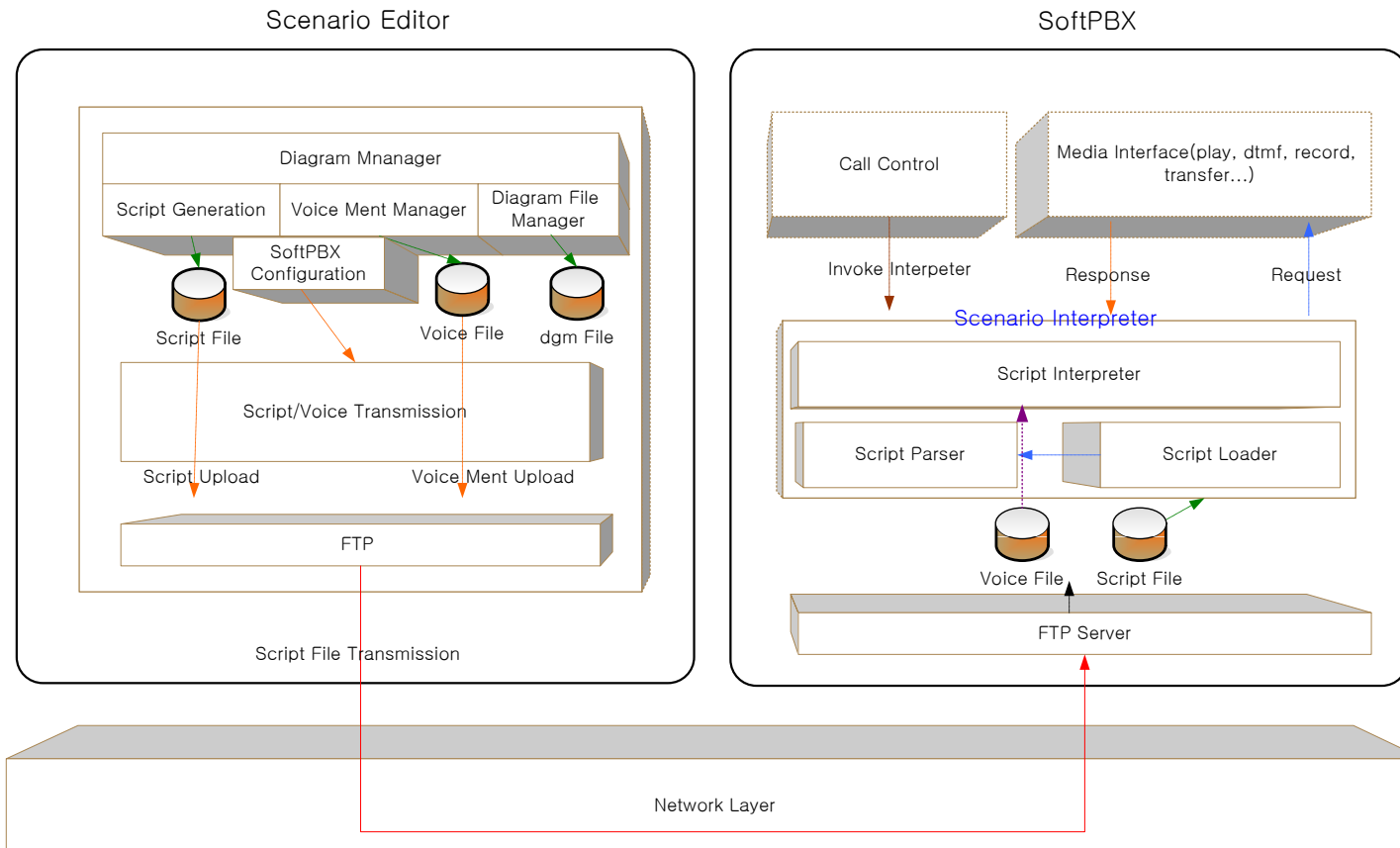
Call Log (Search Condition)

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

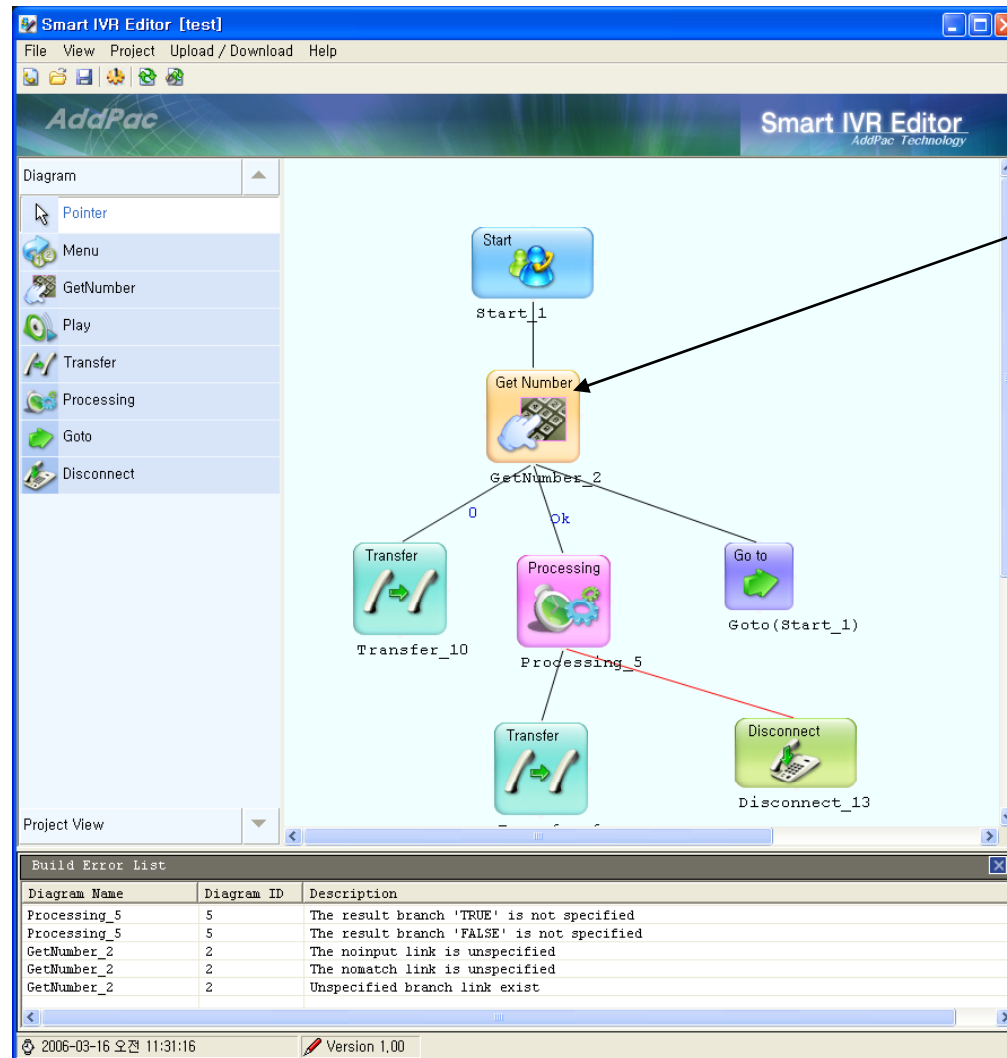


IVR Scenario Editor

IVR Scenario Editor Architecture



IVR Scenario Editor Creation



Component

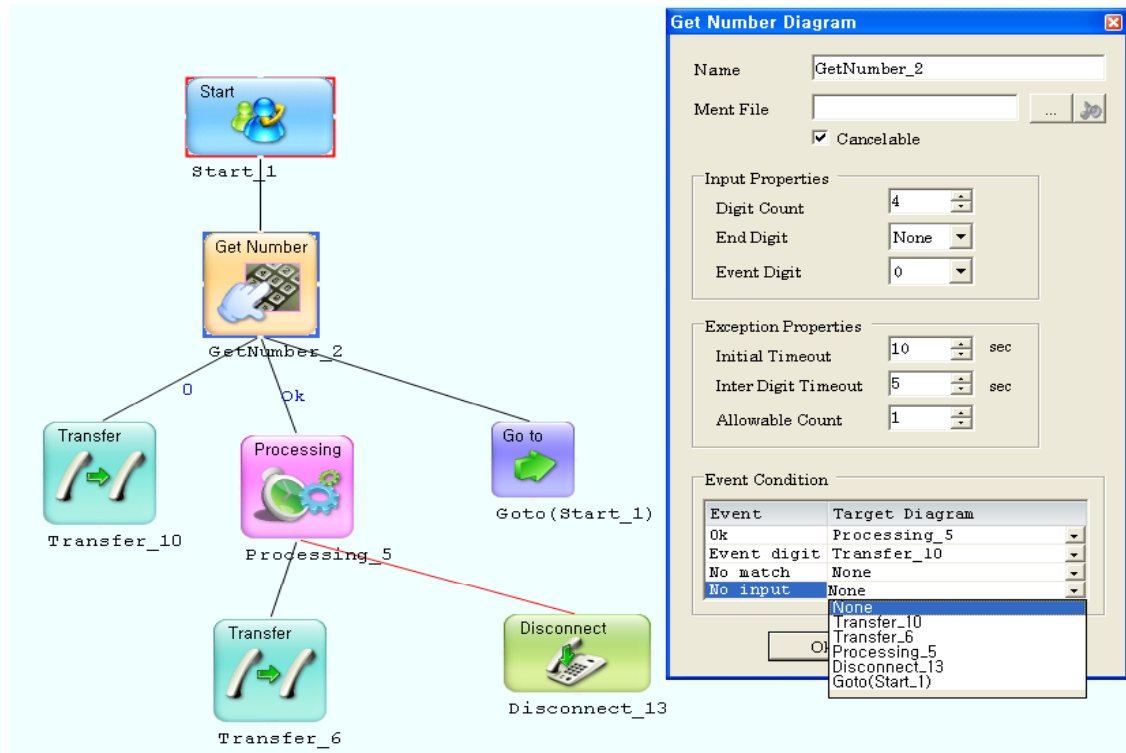
IVR Scenario Editor Creation

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

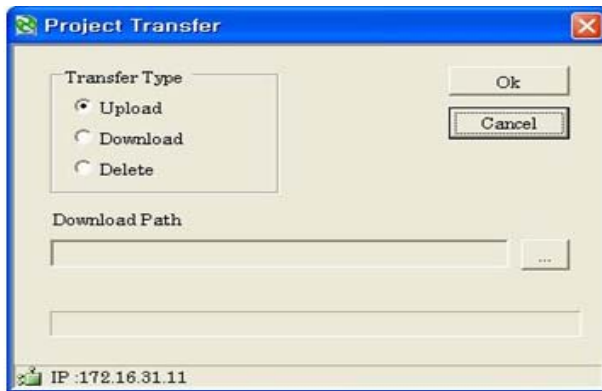
IVR Editor Component Property

IPNext180 IP keyPhone System

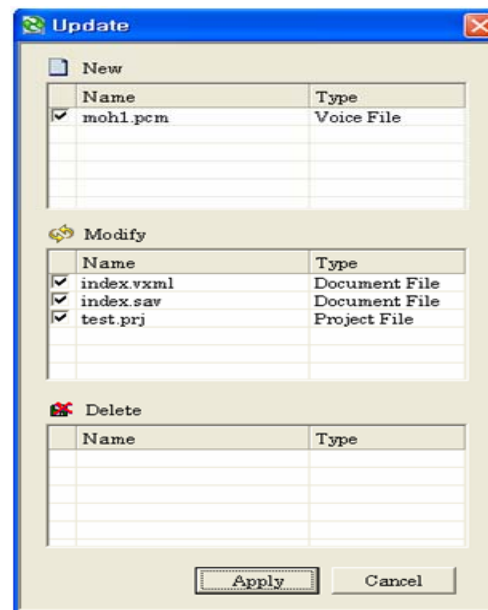


- Support the Property Setting for Component Flow Diagram (Input Event, Exception Properties, Event Condition)
- Provides the different IVR Component Flow depend on Event Condition

IVR Scenario Management



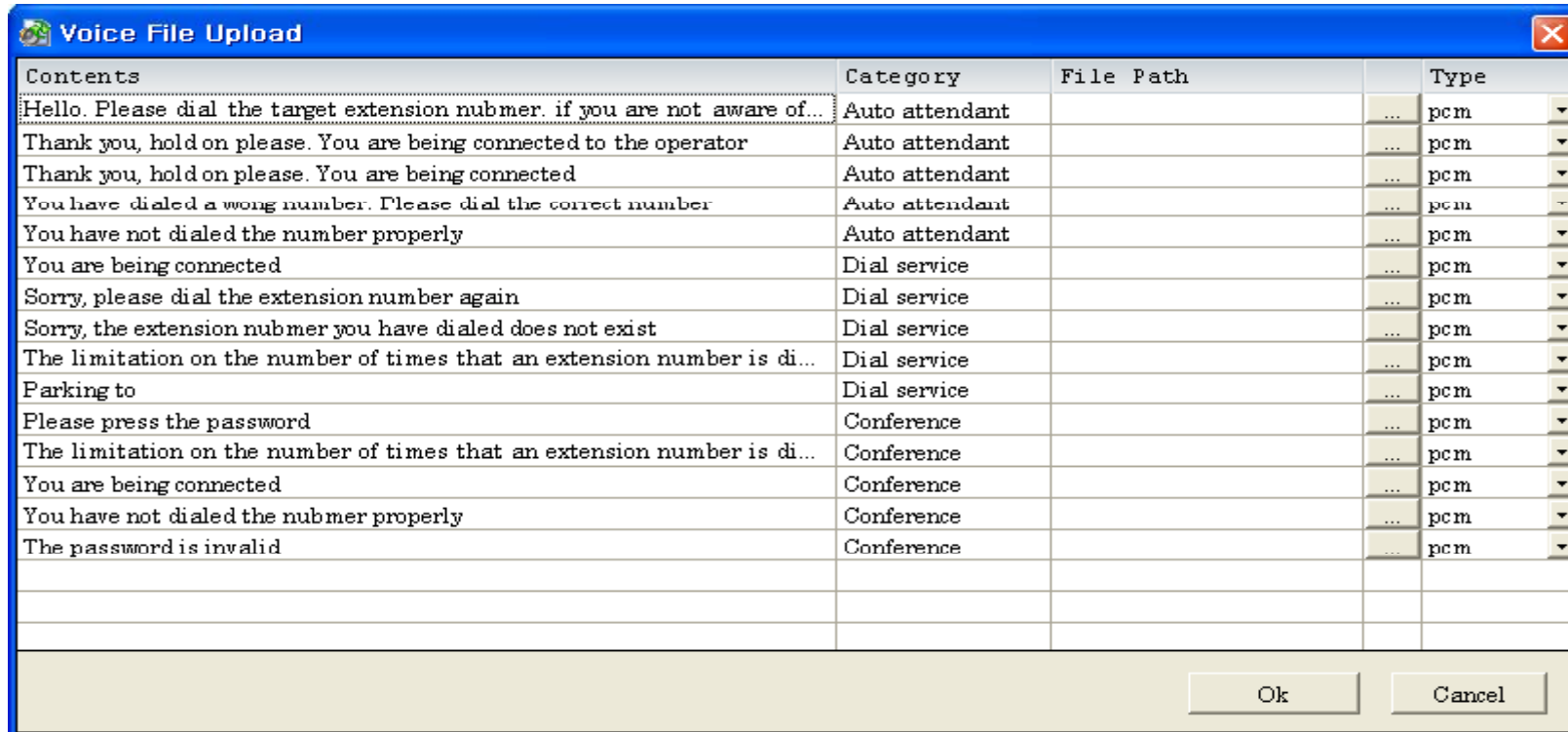
Project Transfer



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



The screenshot shows a window titled "Voice File Upload" with a close button in the top right corner. The window contains a table with four columns: "Contents", "Category", "File Path", and "Type". The "Contents" column lists various voice messages, the "Category" column lists their respective categories, and the "Type" column lists the file format as "pcm". The "File Path" column is currently empty. At the bottom of the window, there are "Ok" and "Cancel" buttons.

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

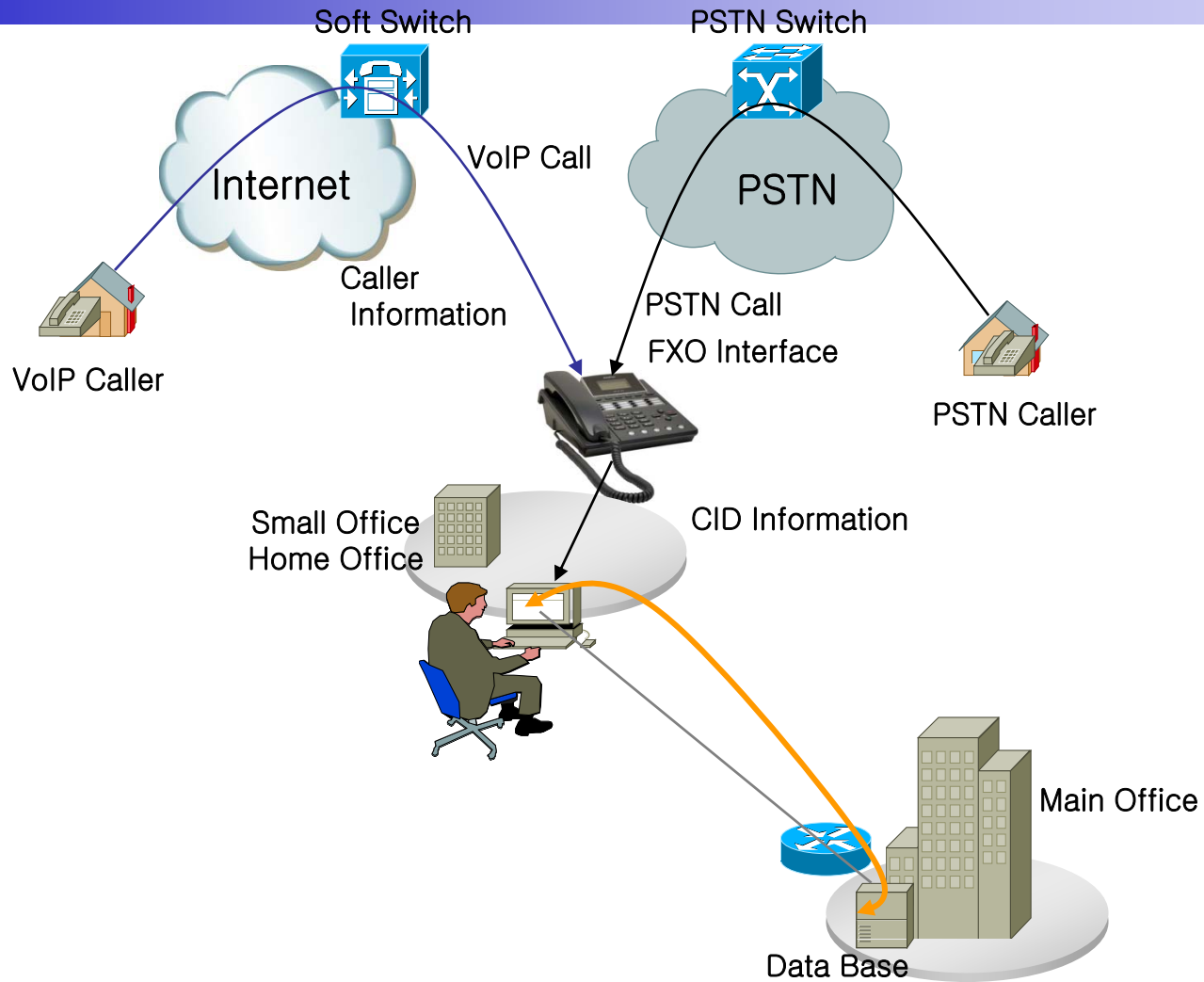


CRM API

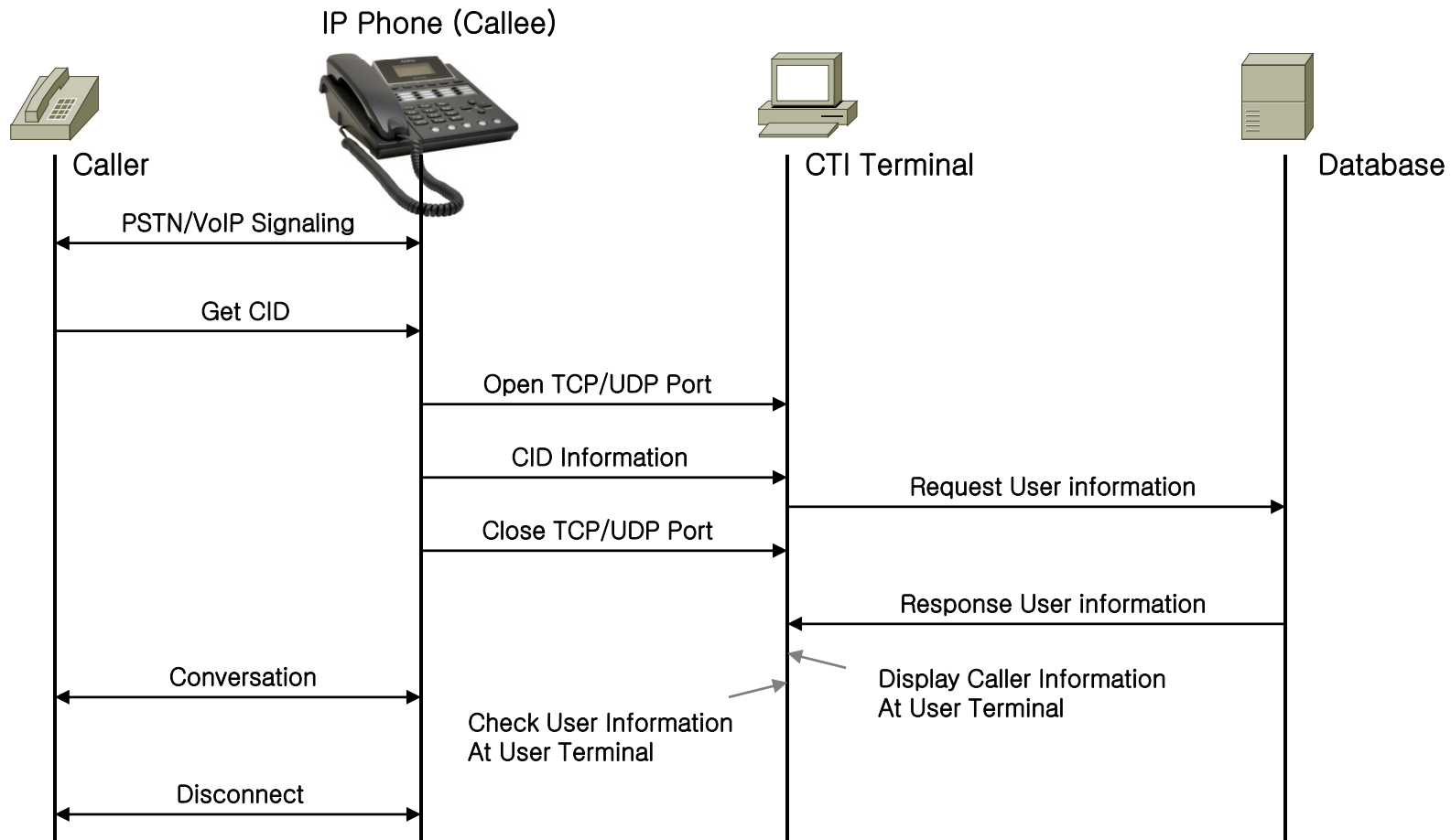
CTI using CID

- 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

Add an Attendant Queue

✓ Add ✗ Cancel ⚙️ Advanced Options

Attendant Queue

Extension * (2~12 digits) Check Extension

Name *

Attendant Members

Extensions	
Extension	<input type="text"/>
Name	Extension

Allowed Attendant Members	
Name	Extension

Description

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.

Related Links

- User Extension
- Address Pool
- Partitions

Attendant Queue Web UI

Advanced Options		
General Settings	Partition	internal
	Address Pool	default
Queue Policy	Queue Size	1024
	Queue Filled 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	<input type="checkbox"/>
	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

Add a Hunt Group

Hunt Group	Extension *	<input type="text"/> (2~12 digits) <input type="button" value="Check Extension"/>
	Name *	<input type="text"/>
	Hunting Mode	Sequential
	No Answer Timeout	10 sec

Group Members	Extensions	Hunt Group Members			
	<table><thead><tr><th>Name</th><th>Extension</th></tr></thead><tbody></tbody></table>	Name	Extension	<table><thead><tr><th>Name</th><th>Extension</th></tr></thead><tbody></tbody></table>	Name
Name	Extension				
Name	Extension				

General Settings	Partition	internal
	Address Pool	default
	Hunt Group Chain	N/A
	Apply Call Forwarding Setting of Members	<input type="checkbox"/>

Description

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.

Related Links

- User Extension
- Partitions
- Address Pool

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!

AddPac Technology Co., Ltd.
Sales and Marketing

Phone +82.2.568.3848 (KOREA)

FAX +82.2.568.3847 (KOREA)

E-mail sales@addpac.com