AddPac VoIP Products

VoIP gateway Training Guide

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Technical Support Team

Phone (02) 568 3848 (2000 ~ 2006)

FAX (02) 568 3847

tech@addpac.com

AddPac Technology

Index

Chapter 1 Introduce AddPac VoIP gateway

Chapter 2 Basic and Advanced Configuration

Chapter 3 Trouble shooting

Chapter 4 Configuration Example

Chapter 1 Introduce AddPac VoIP Gateway

VoIP gateway Function

- AddPac VoIP Gateway Hardware
- AddPac VoIP Gateway feature
- IP Network configuration
- VoIP Network configuration
- Voice Interface configuration

AddPac VoIP Gateway hardware
- (1/2) -

Common Feature

2 of 10 or 100 Base T Ethernet Interface
RJ45 Asynchronous Console Interface

Native Analog Port model – FXS/FXO only

- 1/2 port gateway
  - AP200 Series (Basic)
  - AP300 Series (2 of 100M fast Ethernet)
- 4/8 port gateway
  - AP1000 (FXS4)
  - AP1002 (FXS2O2)
  - AP1005 (FXO8)
  - AP1100(F)
  - AP1200

AddPac VoIP Gateway hardware
- (2/2) -

Slot type model (FXO/FXS/E&M/E1/T1)

Analog (FXS/FXO//E&M)
    1 slot = 4 port (2120 is 1 slot = 8 port)
Digital
    1 slot = 1 port (MG3000 is 1 slot = 2 E1)

1 slot
    AP2110 (1 slot + FXS 4 static)
2 slot
    AP2120 (8 channel *2 slot) – analog only
    AP2520 (Basic)
    AP2620 (2 of 100M fast Ethernet)
    AP2830 (VoIP gateway + ATM router)
4 slot
    AP2850 (VoIP gateway + ATM router)
15 slot
    AP3100 (4channel * 15slot) – analog only

AddPac VoIP Gateway Feature

- WAN/LAN/Routing Service Protocol (1/3) -

WAN/LAN Protocol

- PPPoE Server & Client
- DHCP Server & Client
- 802.1Q VLAN Tagging
- Transparent Bridge for Ethernet
- Transparent Bridge for PPPoE
- PPTP

AddPac VoIP Gateway Feature
- WAN/LAN/Routing Service Protocol (2/3) -

Routing Protocol

- Static Routing
- RIP Version 1, Version 2
- OSPF Version 2
- Source Based Routing Supported

AddPac VoIP Gateway Feature
- WAN/LAN/Routing Service Protocol (3/3) -

Other Scalability

- OoS Based Priority & Bandwidth control
- NTP
- ProxyDNS Server
- DNS Client
- NAT/PAT/IP Share Function
- Stacking
- Auto Upgrade/Configure via VPMS
- IP TOS

AddPac VoIP Gateway Feature

- Management & Security Functions -

Management & Security

- Standard/Extended Access List for Packet filtering
- Telnet/FTP Access Control
- Account List
- Command/Event/Call Logging and support Syslog
- SNMP Agent (MIB V2) and enterprise MIB
- Telnet Server & Client
- FTP Server & Client
- WEB Based Management
- Multi-Level User account management
- Web Base Management

AddPac VoIP Gateway Feature

- Analog Voice Interface –

FXS

- Ring/Tone Generation cadence Configuration Function

  Dial-tone/Reorder-tone/Ring-back-tone/Busy-tone/line-lock

- Polarity inverse Generation
- Caller-ID Generation

  Bell-core/ETSI/ETSI-DTMF

AddPac VoIP Gateway Feature

- Analog Voice Interface –

FXO

- Ring Detect cadence Configuration Function
- Tone Generation cadence Function

  Dial-tone/Ring-back-tone(virtual-ring-back-tone)
- Clear-down-tone(re-order-tone) Function

  Clear-down-tone analyzing Function
- voice confirmed connection Function
- Polarity detect Function
- Caller-ID Generation

  Bell-core/ETSI/ETSI-DTMF

AddPac VoIP Gateway Feature

- Analog Voice Interface–

E&M

- E&M Type 1 ~ 5 supported

  By Jumper Setting (default = 5)
- 2/4 wire method supported

  By Jumper Setting (default = 2 wire)

AddPac VoIP Gateway Feature

- Digital (E1/T1) Voice Interface –

Common

- E1/T1 Supported (different Card)
- LOS (Loss Of Signal) LED
- clock mode Configuration

  internal/external
- PCM-A,PCM-U Configuration

  default : PCMA
- Channel blocking

  Specific channel out-bound block Function
- channel grouping

AddPac VoIP Gateway Feature

- Digital (E1/T1) Voice Interface –

ISDN PRI

- Type ETSI
- Over-lap/En-bloc
- Dial-tone

R2

- MFC/DTMF
- get Call ID (MFC)

AddPac VoIP Gateway Feature
- Voice Over IP Service Protocol –

General Feature 1/2

- G711/G7231/G729/G726 Codec Negotiation
- T38 G3 FAX Relay Protocol
- VAD, CNG(Comfort Noise Generation)
- G.168 echo cancellation
- DTMF Relay

  H245, RTP2833,In-band
- Tone Generation

  Dial-tone/Reorder-tone/Ring-back-tone/Busy-tone

  Clear-down-tone analyzing

  debug rta voice
- Specific call/channel out-bound block function
- Embedded Gate-Keeper

AddPac VoIP Gateway Feature
- Voice Over IP Service Protocol –

General Feature 2/2

- Number Translation
- PLAR(Private Line Auto Ring-down) for hot line
- Call Waiting with FXO
- announcement
- Support Radius Accounting
- programmable H323/SIP signaling port
- Backup and hunting when VoIP Network failed
- make limit conversation time
- VoIP session timer에 의한 call block 방지
- dynamic/static jitter buffer

AddPac VoIP Gateway Feature

- Voice Over IP Service Protocol –

H.323 Feature 1/2

- H.323 V2, 3
- Codec Negotiation
- Direct or GK routed Call Support at same time
- Fast/Slow Start Negotiation
- H245 (No) Tunneling Negotiation
- RAI (Resource Available Indicate)
- announcement
- Support Radius Accounting
- programmable H323/SIP signaling port
- minimize TCP/UDP port for VoIP

AddPac VoIP Gateway Feature

- Voice Over IP Service Protocol –

H.323 Feature 2/2

- controllable OLC channel open timing

  before connect, after h245 opened, after connect
- Selectable H323 alerting message

  Alerting/progress/None

AddPac VoIP Gateway Feature

- Voice Over IP Service Protocol –

SIP Feature 1/2

- SIP Version 2

  RFC3261, RFC2653
- Codec Negotiation
- Direct or Proxy routed Call Support at same time
- Inband/Our of band DTMF Relay

  RTP2833, in-band
- register username or alias(e.164)

  register each dial-peer separately
- register with authentication MD5 hash algorism
- register as Trunk or end-point gateway

AddPac VoIP Gateway Feature
- Voice Over IP Service Protocol –

SIP Feature 2/2

- Call waiting (Call hold)
- Call forwarding

  unconditional/Conditional
- Call transfer

  Attended/blind
- Conference call
- SIP ping for operating behind private Network.

AddPac VoIP Gateway Feature

- Voice Over IP Service Protocol –

MGCP feature

- MGCP Version 1.0

  RFC2705, RFC3435
- Codec Negotiation
- Call-Agent (MGC) Redirection
- Call waiting (Call hold)
- Local Digit-map 저장
- Inband / Message (NTFY) / RTP2833 DTMF-Relay
- Codec, VAD
- MGCP Service port number

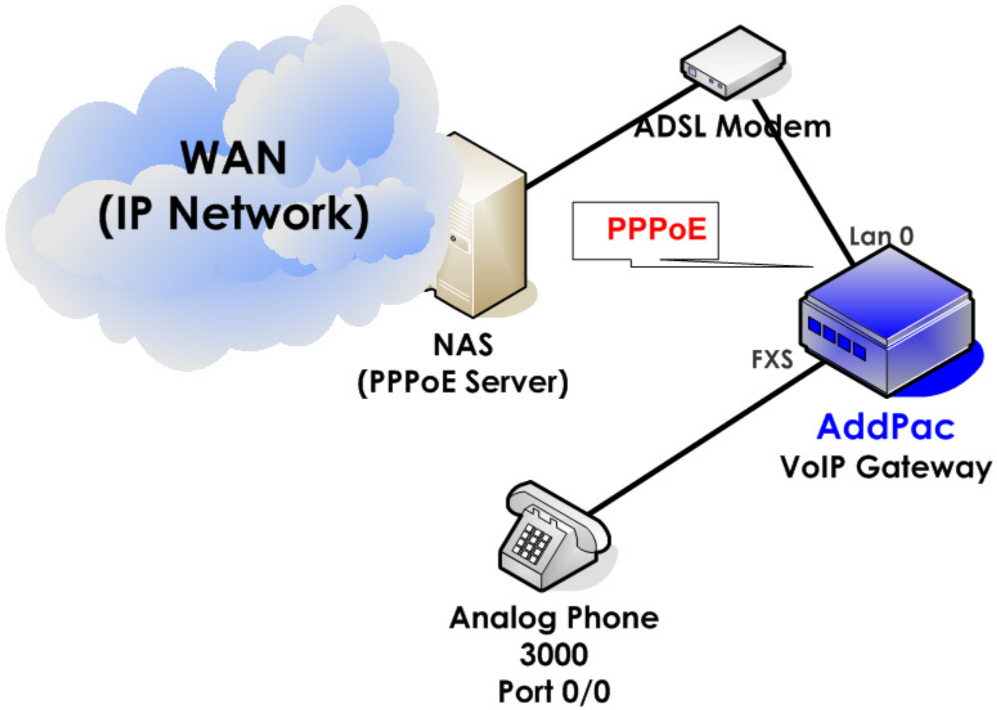
  Local UDP port / Call-Agent (MGC) UDP port

IP Network Configuration

IP Network Configuration

- Dynamic IP via ADSL modem
- Dynamic IP via Cable modem
- Static IP behind Router
- IP Share
- NAT/PAT
- Bridge
- STUN Client

IP Network Configuration
- Dynamic IP via ADSL modem (1/2) –



mandatory setting

LAN0
- **encapsulation PPPoE**
- user ID & password

Pots(VoIP) Peer
- e.164 related configurations

Optional setting

LAN 0
- QoS

LAN 1
- NAT/PAT

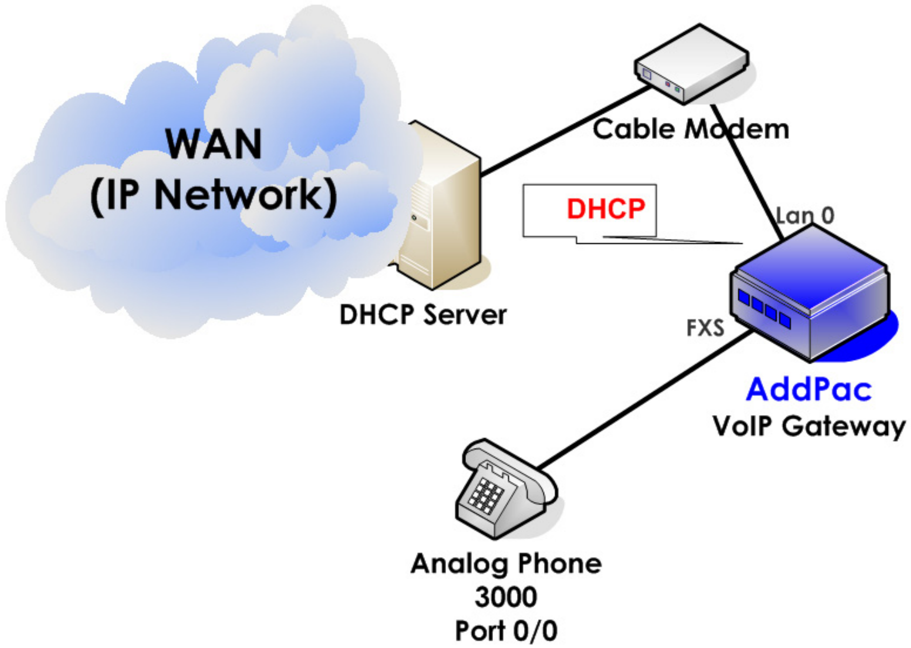
VoIP Network
- GK/SIP/MGCP related configurations

IP Network Configuration
- Dynamic IP via ADSL modem (2/2) –

Configuration Example

```
!
interface ether0.0
 no ip address
 encapsulation pppoe
 ppp authentication pap callin
 ppp pap sent-username addpac password addpac
!
dial-peer voice 0 pots
 destination-pattern 3000
 port 0/0
!
! Voip peer configuration.
!
dial-peer voice 900 voip
 destination-pattern T
 session target 61.222.180.155
…
```

IP Network Configuration
- Dynamic IP via Cable modem (1/2) –



mandatory setting

LAN0
 DHCP Client
Pots(VoIP) Peer
 e.164 related configurations

Optional setting

LAN 0
 QoS
LAN 1
 NAT/PAT
VoIP Network
 GK/SIP/MGCP related configurations

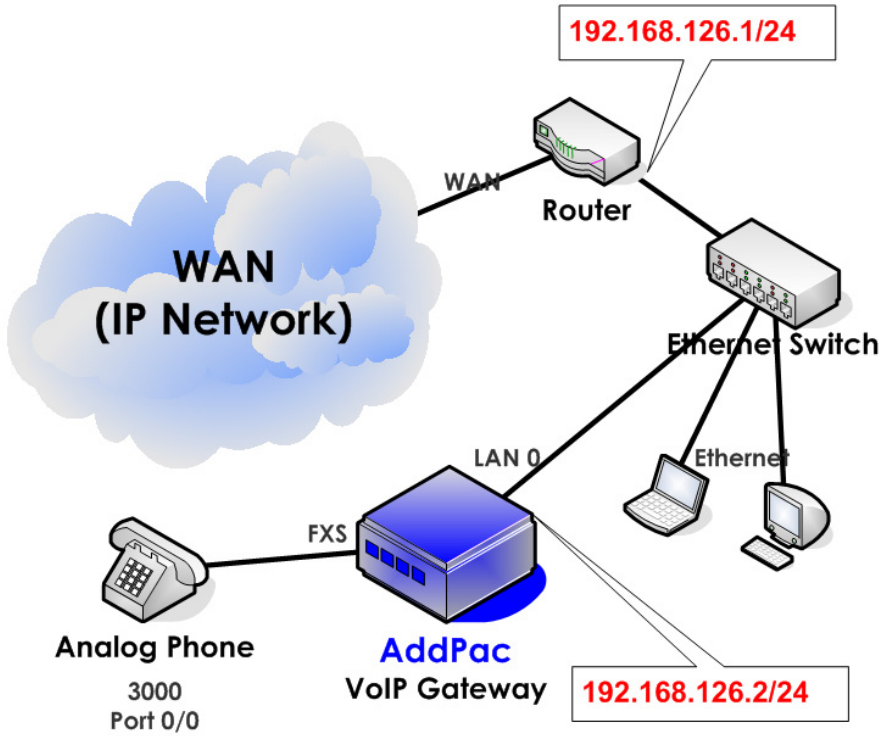
IP Network Configuration

- Dynamic IP via Cable modem (2/2) –

Configuration Example

```
!
interface ether0.0
 ip address dhcp
!
dial-peer voice 0 pots
 destination-pattern 3000
 port 0/0
!
! Voip peer configuration.
!
dial-peer voice 900 voip
 destination-pattern T
 session target 61.222.180.155
…
```

IP Network Configuration
- Static IP behind Router (1/2) –



mandatory setting

LAN0
encapsulation ethernet
ip address
Default routing
Pots(VoIP) Peer
e.164 related configurations

Optional setting

LAN 0
QoS
LAN 1
NAT/PAT
VoIP Network
GK/SIP/MGCP related configurations

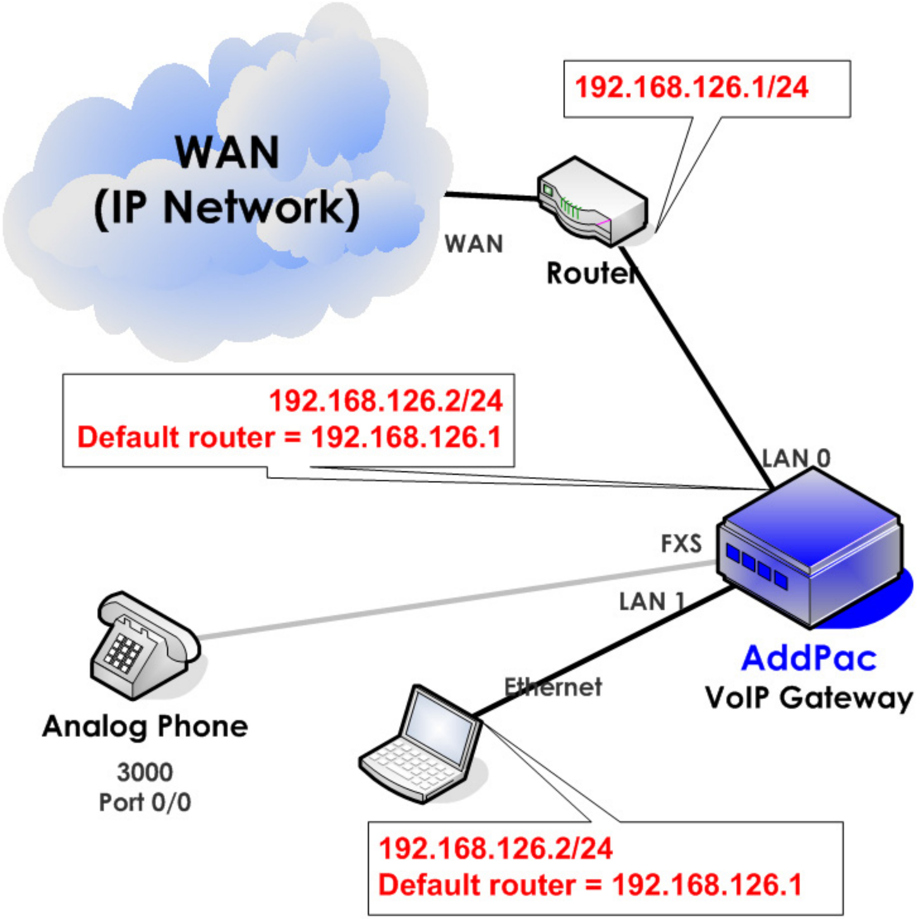
IP Network Configuration

- Static IP behind Router (2/2) –

Configuration Example

```
!
interface ether0.0
 ip address 192.168.126.2 255.255.255.0
!
Route 0.0.0.0 0.0.0.0 192.169.126.1
!
dial-peer voice 0 pots
 destination-pattern 3000
 port 0/0
!
! Voip peer configuration.
!
dial-peer voice 900 voip
 destination-pattern T
 session target 61.222.180.155
…
```

IP Network Configuration
- IP Share (1/7) –



mandatory setting

LAN0
Dynamic Or Static IP
LAN 1
PPPoE/DHCP Server
or No IP (Static)
Default routing (Static)
IP Share
Default routing
Pots(VoIP) Peer
e.164 related configurations

Optional setting

LAN 0
QoS
LAN 1
NAT/PAT can not be configured
VoIP Network
GK/SIP/MGCP related configurations

IP Network Configuration

- IP Share (2/7) –

IP Share Configuration Table MAP

Up-Link Interface (LAN 0)	Local Interface (LAN 1)	Configuration Map
DHCP DHCP Client	DHCP (Server)	O (1)
	PPPoE (Server)	O (2)
	Static (No IP)	X
PPP PPPoE Client	DHCP	O (3)
	PPPoE	O (4)
	Static	X
Static	DHCP	X
	PPP	X
	Static	O (5)

IP Network Configuration

- IP Share (3/7) –

Configuration Example 1 (DHCP to DHCP)

```
!
dhcp-list 0 address server interface ether0.0
dhcp-list 0 option dhcp-lease-time 600
!
ip-share enable
ip-share interface net-side ether0.0
ip-share interface local-side ether1.0
!
interface ether0.0
ip address dhcp
qos 200 150
!
interface ether1.0
 no ip address
 ip dhcp-group 0
!
```

IP Network Configuration

- IP Share (4/7) –

Configuration Example 2 (DHCP to PPPoE)

```
!
ip-share enable
ip-share interface net-side ether0.0
ip-share interface local-side ether1.0
!
interface ether0.0
ip address dhcp
qos 200 150
!
interface ether1.0
  no ip address
  encapsulation pppoe
  ppp authentication pap callin
  ppp pap sent-username addpac password test
  ppp echo interval 20
  ppp ipcp ms-dns
  ppp ipcp default-route
 ppp role server
!
```

IP Network Configuration

- IP Share (5/7) –

Configuration Example 3 (PPPoE to DHCP)

```
!
dhcp-list 0 address server interface ether0.0
dhcp-list 0 option dhcp-lease-time 600
!
ip-share enable
ip-share interface net-side ether0.0
ip-share interface local-side ether1.0
!
interface ether0.0
  no ip address
  encapsulation pppoe
  ppp authentication pap callin
  ppp pap sent-username addpac password test
  ppp echo interval 20
  ppp ipcp ms-dns
  ppp ipcp default-route
  qos 200 150
!
interface ether1.0
  no ip address
  ip dhcp-group 0
!
```

IP Network Configuration

- IP Share (6/7) –

Configuration Example 4 (PPPoE to PPPoE)

```
ip-share enable
ip-share interface net-side ether0.0
ip-share interface local-side ether1.0
!
interface ether0.0
  no ip address
  encapsulation pppoe
  ppp authentication pap callin
  ppp pap sent-username addpac password test
  ppp echo interval 20
  ppp ipcp ms-dns
  ppp ipcp default-route
  qos 200 150
!
interface ether1.0
   no ip address
   encapsulation pppoe
   ppp authentication pap callin
   ppp pap sent-username addpac password test
   ppp echo interval 20
   ppp ipcp ms-dns
   ppp ipcp default-route
  ppp role server
```

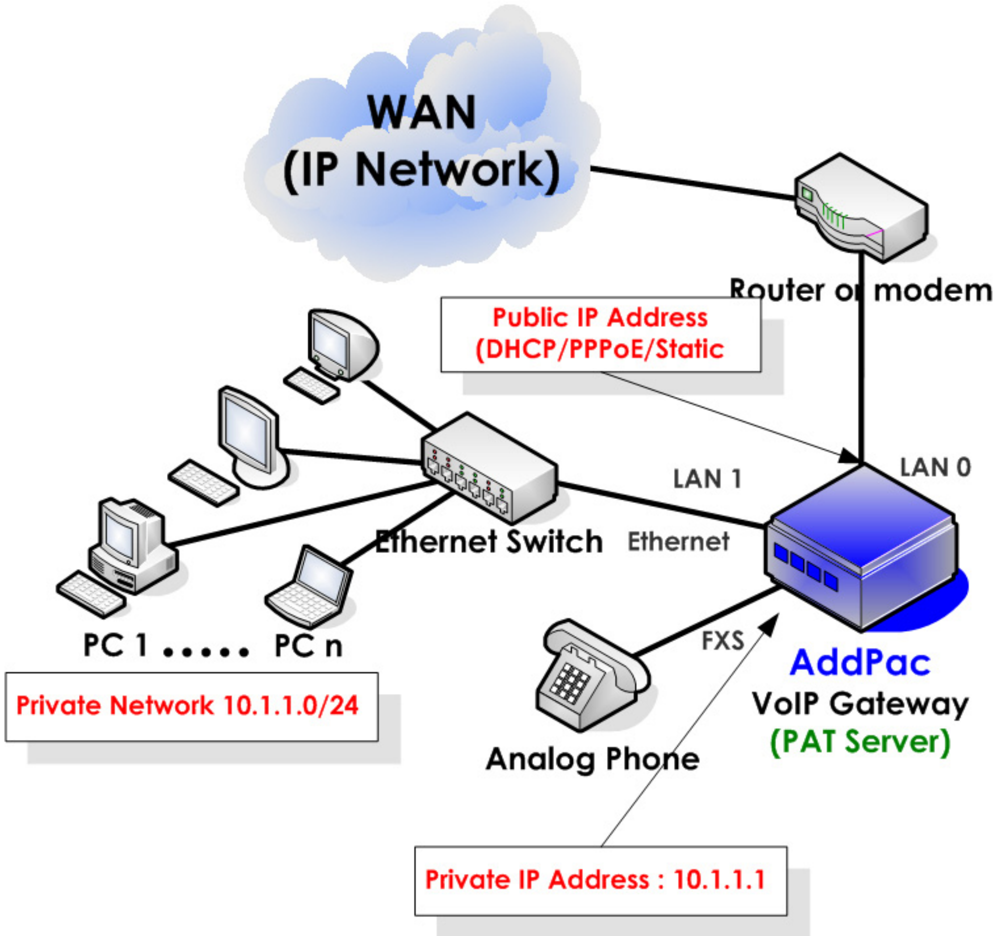
IP Network Configuration

- IP Share (7/7) –

Configuration Example 5 (Static to Static)

```
!
ip-share enable
ip-share interface net-side ether0.0
ip-share interface local-side ether1.0
!
interface ether0.0
 ip address 192.168.126.2 255.255.255.0
!
interface ether1.0
 no ip address
!
route 0.0.0.0 0.0.0.0 192.168.126.1
!
```

IP Network Configuration
- NAT/PAT (1/2) –



mandatory setting

LAN0
Dynamic Or Static IP
LAN 1
Private IP
Binding PAT pool
PAT pool
Default routing (Static)
or No IP (Static)
Default routing (Static)
Default routing
Pots(VoIP) Peer
e.164 related configurations

Optional setting

LAN 0
QoS
VoIP Network
GK/SIP/MGCP related
configurations

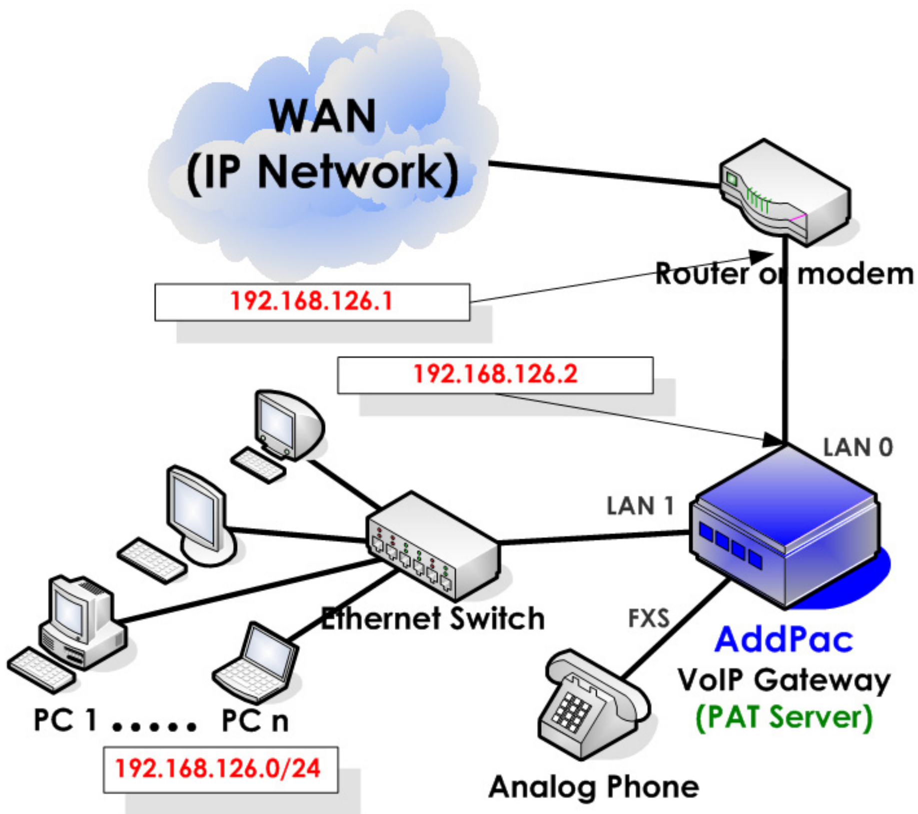
IP Network Configuration

- NAT/PAT (2/2) –

Configuration Example

```
!
nat-list 1 pat static-entry tcp 1720  local
nat-list 1 pat group-static-entry udp 22000 30000 local
nat-list 1 pat group-static-entry tcp 10000 22000 local
nat-list 1 pat static-entry tcp 23  local
nat-list 1 pat group-static-entry tcp 20 21 local
nat-list 1 pat group-static-entry udp 67 68 local
nat-list 1 pat static-entry icmp ping  local
!
interface ether0.0
 ip address dhcp
 qos-control 200 100
!
interface ether1.0
 ip address 10.1.1.1 255.255.255.0
 ip nat-group 1 pat ether0.0
```

IP Network Configuration
- Bridge (1/2) –



mandatory setting

LAN0
- **IP address (Static IP only)**
- **bridge**

LAN 1
- **no IP address**
- **bridge**

Default routing

no ip routing

No Bridge Spanning Tree

Pots(VoIP) Peer
- e.164 related configurations

Optional setting

LAN 0
- QoS

LAN 1
- **NAT/PAT can not be configured**

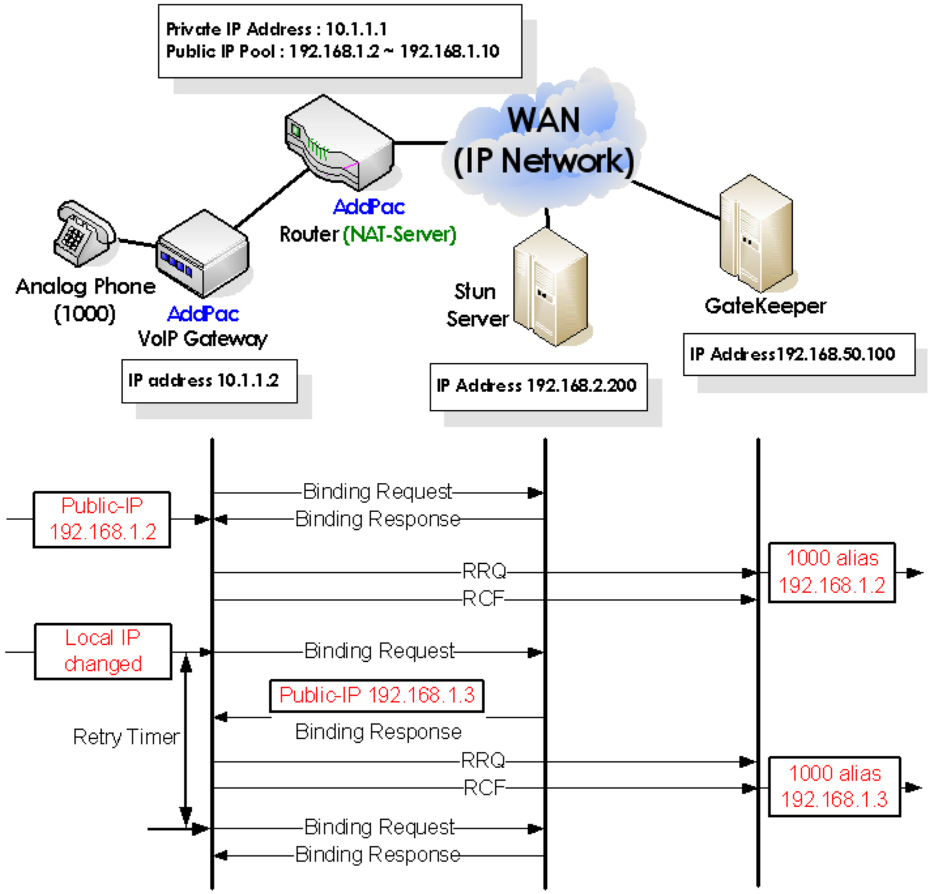
VoIP Network
- GK/SIP/MGCP related configurations

IP Network Configuration
- Bridge (2/2) –

Configuration Example

```
!
No ip routing
interface ether0.0
 ip address 192.168.126.2 255.255.255.0
 bridge
!
interface ether1.0
 no ip address
 bridge
!
No bridge spanning tree
route 0.0.0.0 0.0.0.0 192.168.126.1
!
```

IP Network Configuration
- STUN client –



Command

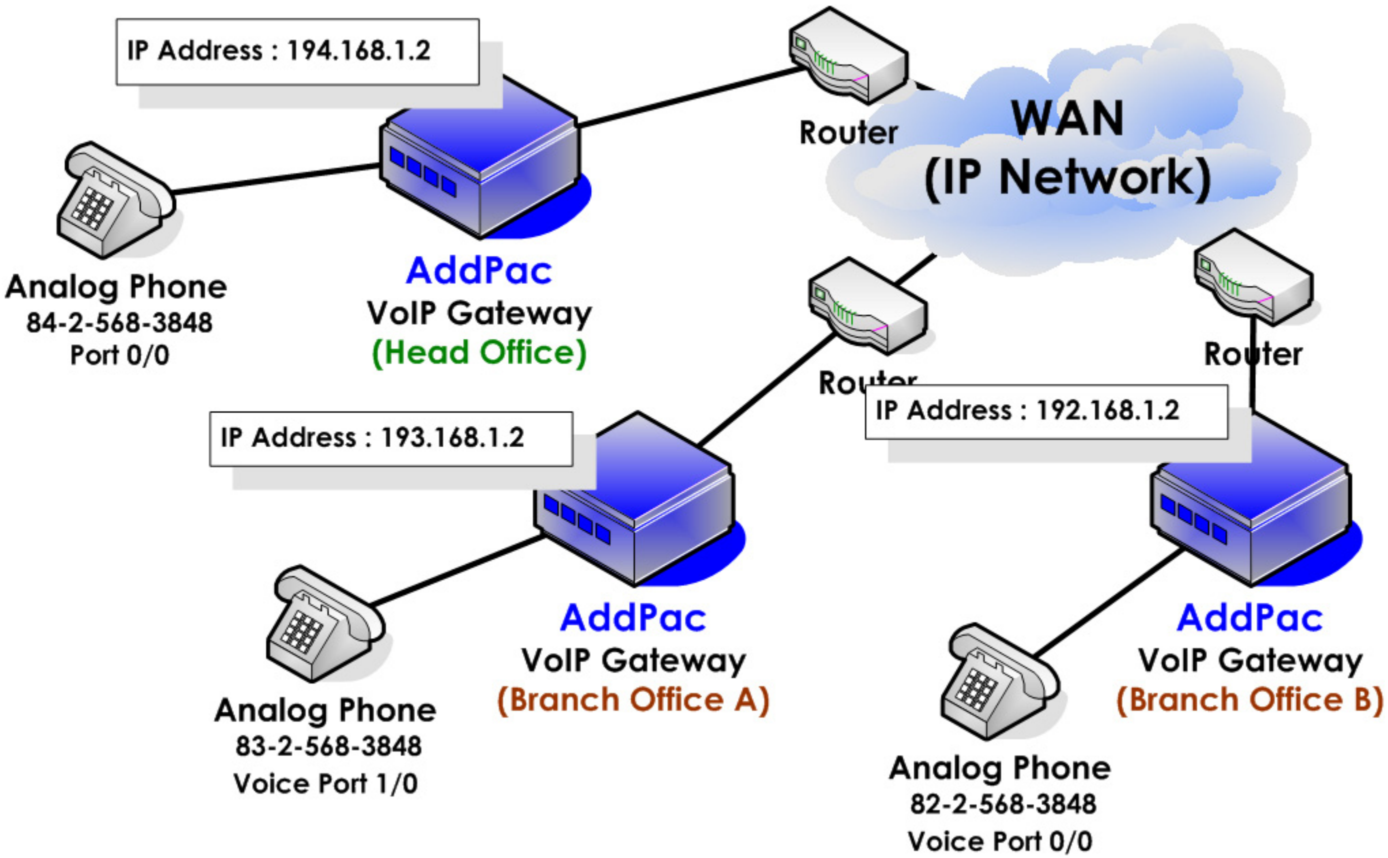
```
config
  interface ether 0.0
    ip stun-server 61.33.161.111
    ip stun-server retry-time 10
    ip stun-server (default = disabled)
```

VoIP Network Configuration

VoIP Network 설정

- Direct Call with H323/SIP
- H323 GK routed Call
- SIP Proxy routed Call
- SIP Related configuration

VoIP Network Configuration
- direct Call A(1/3)–



VoIP Network Configuration

- direct Call A (2/3)–

H323

Head Office

```
!
hostname HO
interface ether0.0 ether0.0
 ip address 194.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
destination-pattern 8425683848
port 0/0
!
dial-peer voice 1000 voip
destination-pattern 8325683848
session target 193.168.1.2
!
dial-peer voice 1001 voip
  destination-pattern 82T
 session target 192.168.1.2
```

Branch Office A

```
!
hostname A
interface ether0.0
 ip address 193.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 193.168.1.1
!
dial-peer voice 0 pots
 destination-pattern 8325683848
 port 1/0
!
dial-peer voice 1000 voip
 destination-pattern 82........
  session target 192.168.1.2
!
dial-peer voice 1001 voip
  destination-pattern 8425683848
 session target 194.168.1.2
```

VoIP Network Configuration
- direct Call A (3/3)–

SIP

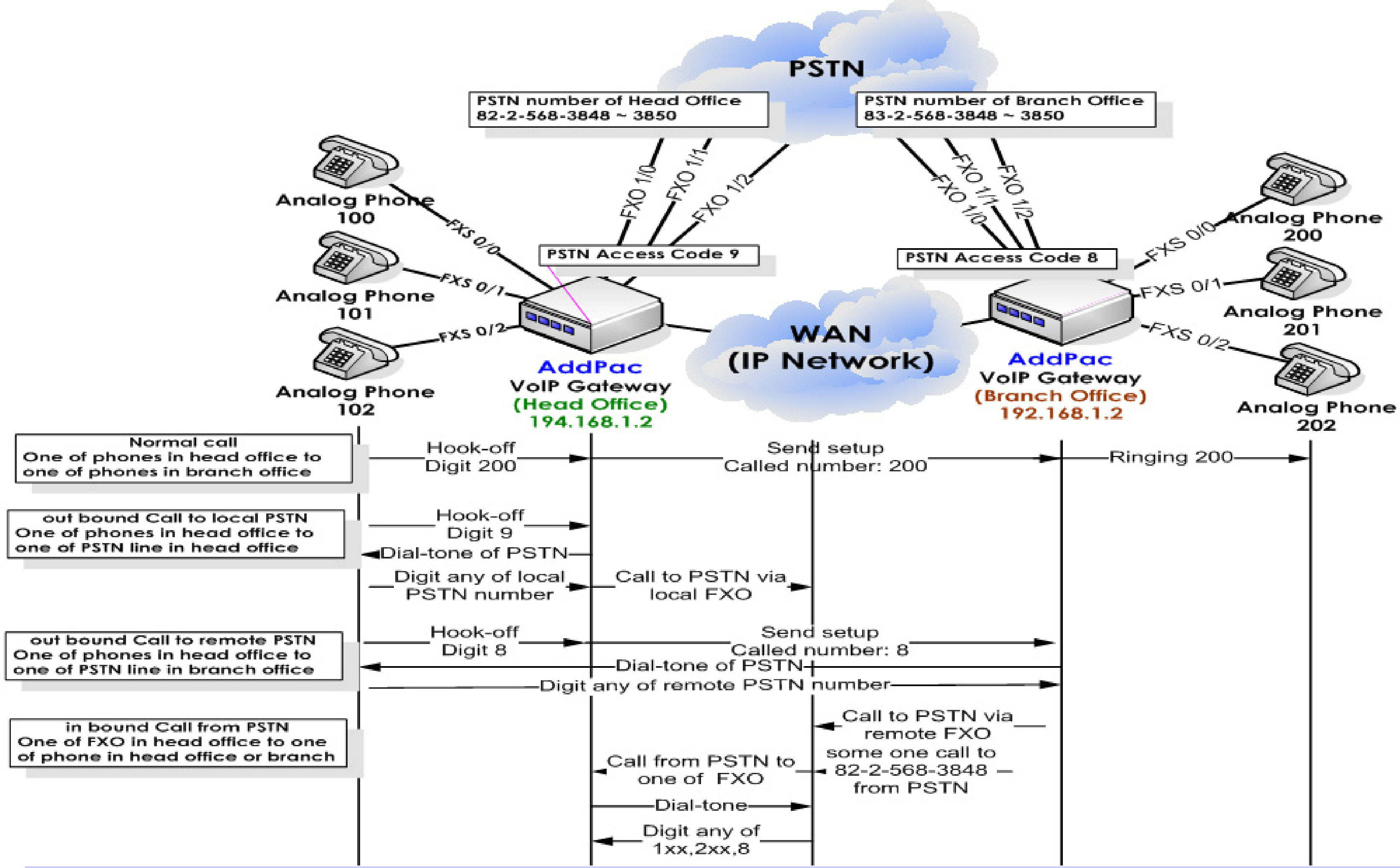
Head Office

```
!
hostname HO
interface ether0.0
 ip address 194.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
! Pots peer configuration.
!
dial-peer voice 0 pots
 destination-pattern 8425683848
 port 0/0
!
dial-peer voice 1000 voip
 destination-pattern 8325683848
session target 193.168.1.2
session protocol sip
dtmf-relay rtp-2833
```

Branch Office A

```
!
hostname A
interface ether0.0
 ip address 193.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 193.168.1.1
!
! Pots peer configuration.
!
dial-peer voice 0 pots
 destination-pattern 8325683848
 port 1/0
!
dial-peer voice 1000 voip
 destination-pattern 82……..
 session target 192.168.1.2
session protocol sip
dtmf-relay rtp-2833
```

VoIP Network Configuration
- direct Call B (1/3)–



VoIP Network Configuration

- direct Call B (2/3)–

H323

Head Office

```
!
hostname HO
interface ether0.0 ether0.0
 ip address 194.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
destination-pattern 100
port 0/0
!
dial-peer voice 0 pots
destination-pattern 101
port 0/1
!
dial-peer voice 0 pots
destination-pattern 102
port 0/2
!
dial-peer voice 0 pots
destination-pattern 9
port 1/0
!
dial-peer voice 0 pots
destination-pattern 9
port 1/1
!
dial-peer voice 0 pots
destination-pattern 9
port 1/2
!
dial-peer voice 1000 voip
destination-pattern 2..
session target 192.168.1.2
!
dial-peer voice 1000 voip
destination-pattern 8
session target 192.168.1.2
```

Branch Office

```
!
hostname HO
interface ether0.0 ether0.0
 ip address 192.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
destination-pattern 200
port 0/0
!
dial-peer voice 0 pots
destination-pattern 201
port 0/1
!
dial-peer voice 0 pots
destination-pattern 202
port 0/2
!
dial-peer voice 0 pots
destination-pattern 8
port 1/0
!
dial-peer voice 0 pots
destination-pattern 8
port 1/1
!
dial-peer voice 0 pots
destination-pattern 8
port 1/2
!
dial-peer voice 1000 voip
destination-pattern 1..
session target 194.168.1.2
!
dial-peer voice 1000 voip
destination-pattern 9
session target 194.168.1.2
```

VoIP Network Configuration

- direct Call B (3/3)–

SIP

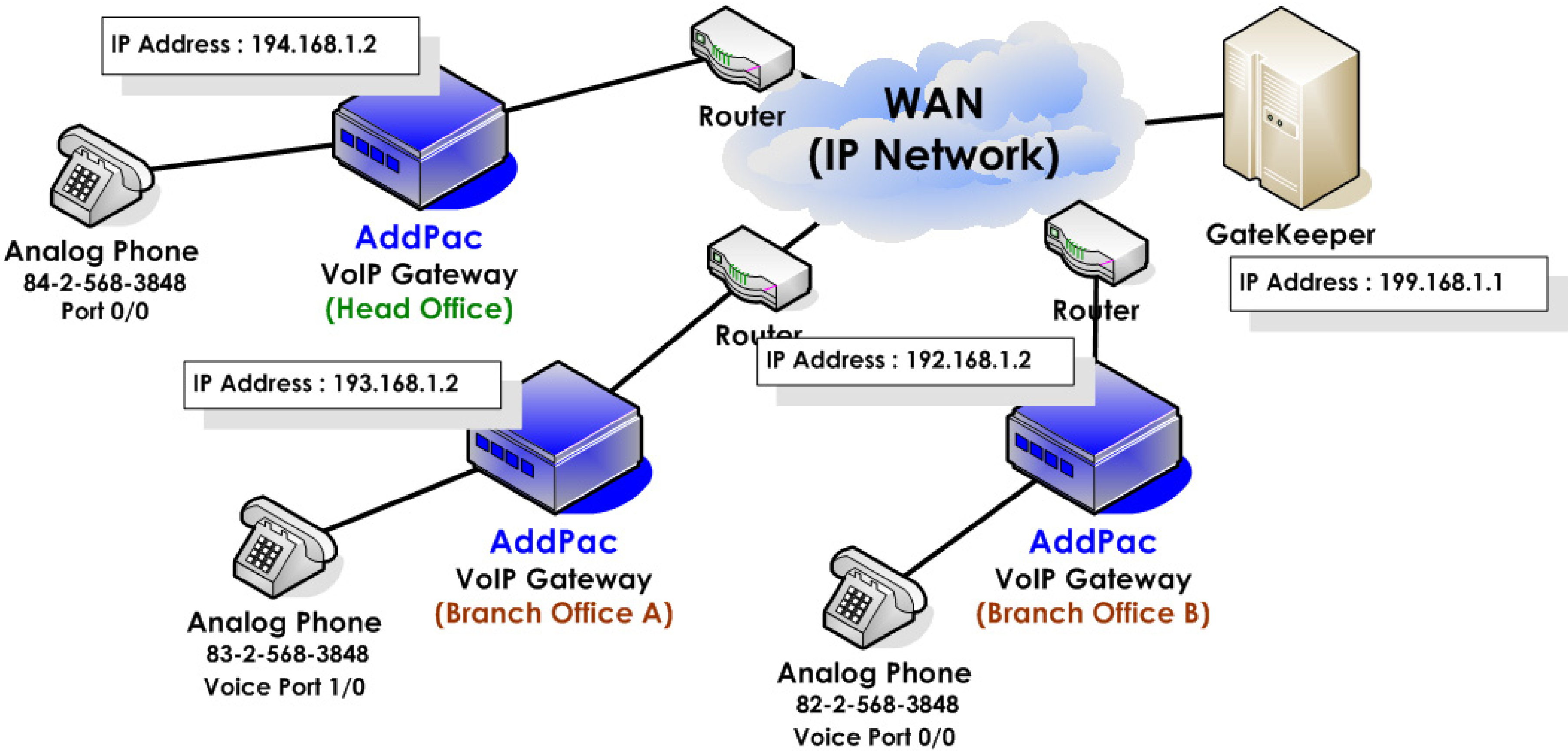
Head Office

```
!
hostname HO
interface ether0.0 ether0.0
 ip address 194.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
destination-pattern 100
port 0/0
!
dial-peer voice 0 pots
destination-pattern 101
port 0/1
!
dial-peer voice 0 pots
destination-pattern 102
port 0/2
!
dial-peer voice 0 pots
destination-pattern 9
port 1/0
!
dial-peer voice 0 pots
destination-pattern 9
port 1/1
!
dial-peer voice 0 pots
destination-pattern 9
port 1/2
!
dial-peer voice 1000 voip
destination-pattern 2..
session protocol sip
session target 192.168.1.2
!
dial-peer voice 1000 voip
destination-pattern 8
session protocol sip
session target 192.168.1.2
```

Branch Office

```
!
hostname HO
interface ether0.0 ether0.0
 ip address 192.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
destination-pattern 200
port 0/0
!
dial-peer voice 0 pots
destination-pattern 201
port 0/1
!
dial-peer voice 0 pots
destination-pattern 202
port 0/2
!
dial-peer voice 0 pots
destination-pattern 8
port 1/0
!
dial-peer voice 0 pots
destination-pattern 8
port 1/1
!
dial-peer voice 0 pots
destination-pattern 8
port 1/2
!
dial-peer voice 1000 voip
destination-pattern 1..
session protocol sip
session target 194.168.1.2
!
dial-peer voice 1000 voip
destination-pattern 9
session protocol sip
session target 194.168.1.2
```

VoIP Network Configuration
- H323 GK routed Call (1/2) –



VoIP Network Configuration
- H323 GK routed Call (2/2) –

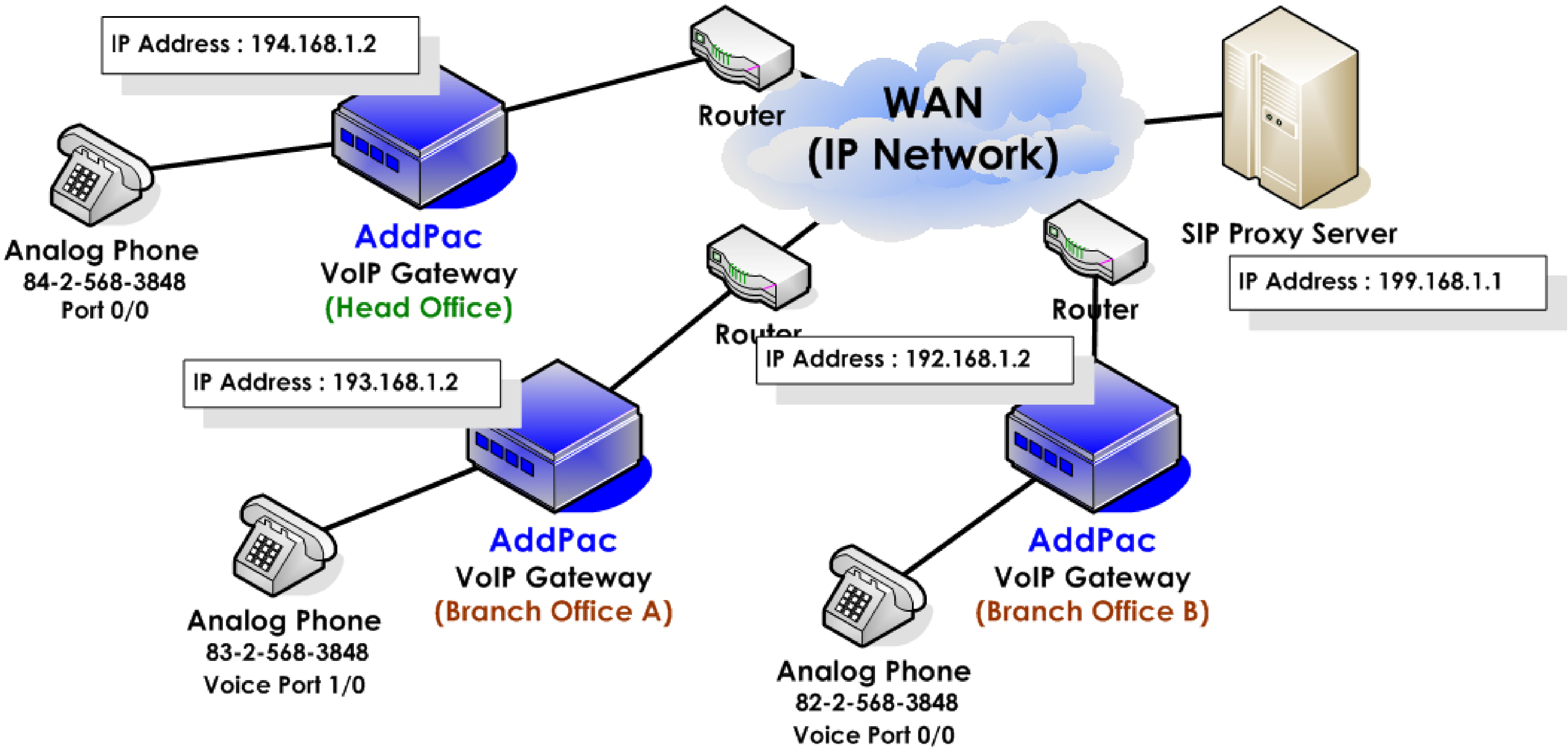
Head Office

```
!
hostname HO
interface ether0.0
 ip address 194.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
destination-pattern 8425683848
port 0/0
!
dial-peer voice 1000 voip
destination-pattern 8325683848
session target ras
!
gateway
 h323-id addpac-HQ
gkip 199.168.1.1 1719 128
register
```

Branch Office A

```
!
hostname A
interface ether0.0
 ip address 193.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
 destination-pattern 8325683848
 port 1/0
!
dial-peer voice 1000 voip
destination-pattern T
session target ras
!
gateway
 h323-id addpac-A
gkip 199.168.1.1 1719 128
register
```

VoIP Network Configuration
- SIP Proxy routed Call (1/2) –



VoIP Network Configuration

- SIP Proxy routed Call (2/2) –

Head Office

```
hostname HO
interface ether0.0
 ip address 194.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 194.168.1.1
!
dial-peer voice 0 pots
 destination-pattern 8425683848
 port 0/0
!
dial-peer voice 1000 voip
 destination-pattern T
 session target sip-server
session protocol sip
dtmf-relay rtp-2833
!
sip-ua
 sip-username 8225683848
sip-password AddPac-HO
sip-server 199.168.1.1
register e164
```

Branch Office A

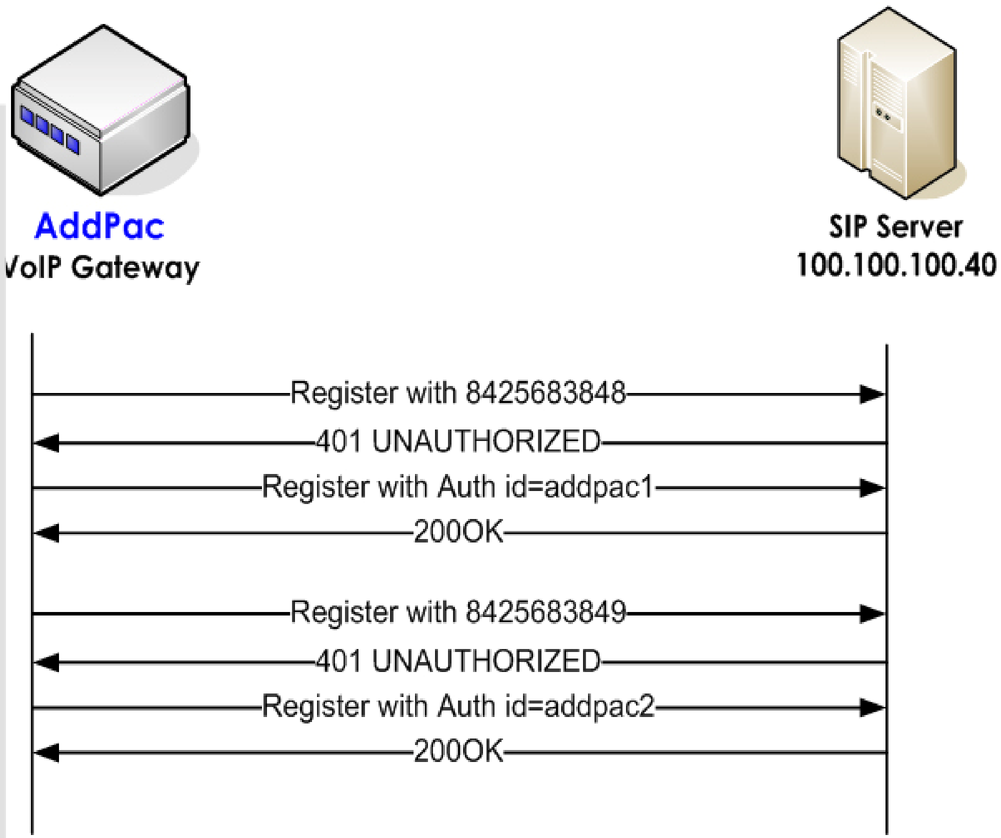
```
hostname A
interface ether0.0
 ip address 193.168.1.2 255.255.255.0
!
route 0.0.0.0 0.0.0.0 193.168.1.1
!
dial-peer voice 0 pots
 destination-pattern 8225683848
 port 0/0
!
dial-peer voice 1000 voip
 destination-pattern T
 session target sip-server
session protocol sip
dtmf-relay rtp-2833
!
sip-ua
 sip-username 8425683848
sip-password AddPac-A
 sip-server 199.168.1.1
 register e164
```

VoIP Network Configuration
- SIP Proxy Related Configuration –

Dial-peer Register(1/2)

Configuration

```
dial-peer voice 0 pots
 destination-pattern 8425683848
 port 0/0
 user-name addpac1
 user-password addpac1
!
dial-peer voice 1 pots
 destination-pattern 8425683849
 port 0/0
 user-name addpac2
 user-password addpac2
!
Sip-ua
 sip-server 100.100.100.40
 register e164
```



VoIP Network Configuration

- SIP Proxy Related Configuration –

Dial-peer Register(1/2)

Sending SIP PDU to (100.1.1.40:5060) from 5060
REGISTER sip:100.1.1.40 SIP/2.0
Via:SIP/2.0/100.100.100.1:5060;branch=z9hG4bK5543002ea434
From: <sip:8225683848@100.1.1.40>;tag=5543002ea4
To: sip:8225683848@100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Contact: <sip:8225683848@100.1.1.2>;expires=3600
Expires: 3600
Content-Length: 0
Max-Forwards: 70

Received SIP PDU from (100.100.100.40:5060)
SIP/2.0 401 UNAUTHORIZED
To: sip:8225683848@100.1.1.40;tag=10efb2232
From: <sip:8225683848@100.1.1.40>;tag=5543002ea4
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea434
WWW-Authenticate: Digest realm="100.1.1.40", nonce="33f16bf255ec473dd45911db0d0a9dc1", algorithm=MD5

Sending SIP PDU to (100.1.1.40:5060) from 5060
REGISTER sip:registrar.100.1.1.40 SIP/2.0
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea435
From: <sip:8225683848@100.1.1.40>;tag=5543002ea4
To: sip:8225683848@100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 35 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Authorization: Digest username="addpac1", realm="100.1.1.40", nonce="33f16bf255ec473dd45911db0d0a9dc1", uri=".100.1.1.40", respon se="8b72ee2edd93765a8b6ae8896fcb4136", algorithm=MD5
Contact: <sip:8225683848@100.1.1.2>;expires=3600
Expires: 3600
Content-Length: 0
Max-Forwards: 70

Received SIP PDU from (100.100.100.40:5060)
SIP/2.0 200 OK
P-Associated-Uri: <sip:8225683848@100.1.1.40>
To: sip:8225683848@100.1.1.40;tag=2ae655e
From: <sip:8225683848@100.1.1.40>;tag=5543002ea4
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 35 REGISTER

REGISTER sip:100.1.1.40 SIP/2.0
Via:SIP/2.0/100.100.100.1:5060;branch=z9hG4bK5543002ea434
From: <sip:8225683849@100.1.1.40>;tag=5543002ea4
To: sip:8225683848@100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Contact: <sip:8225683848@61.33.161.71>;expires=3600
Expires: 3600
Content-Length: 0
Max-Forwards: 70

SIP/2.0 401 UNAUTHORIZED
To: sip:8225683848@100.1.1.40;tag=10efb2232
From: <sip:8225683848@100.1.1.40>;tag=5543002ea4
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea434
WWW-Authenticate: Digest realm="100.1.1.40", nonce="33f16bf255ec473dd45911db0d0a9dc1", algorithm=MD5

Sending SIP PDU to (100.1.1.40:5060) from 5060
REGISTER sip:registrar.100.1.1.40 SIP/2.0
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea435
From: <sip:8225683848@100.1.1.40>;tag=5543002ea4
To: sip:8225683848@100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 35 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Authorization: Digest username="addpac2", realm="100.1.1.40",

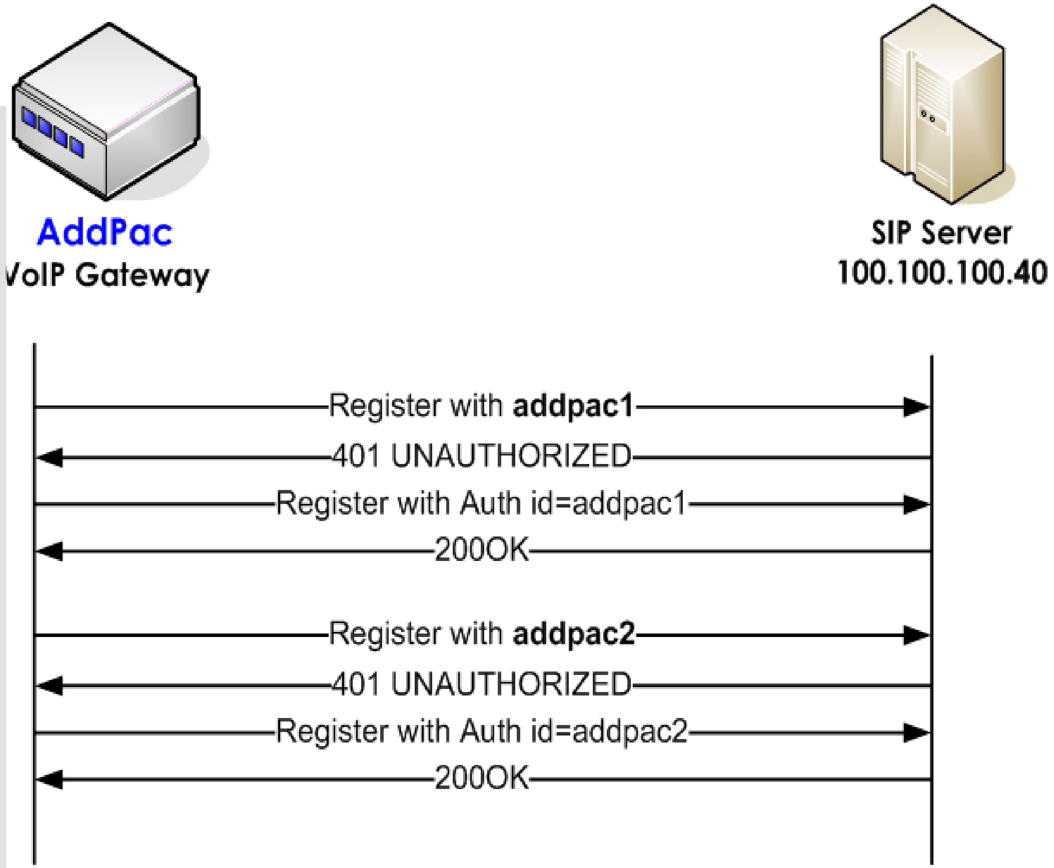
VoIP Network Configuration

- SIP Proxy Related Configuration –

user Register(1/2)

Configuration

```
dial-peer voice 0 pots
 destination-pattern 8425683848
 port 0/0
 user-name addpac1
 user-password addpac1
!
dial-peer voice 1 pots
 destination-pattern 8425683849
 port 0/0
 user-name addpac2
 user-password addpac2
!
Sip-ua
 user-register
 sip-server 100.100.100.40
 register e164
```



VoIP Network Configuration

- SIP Proxy Related Configuration –

user Register(2/2)

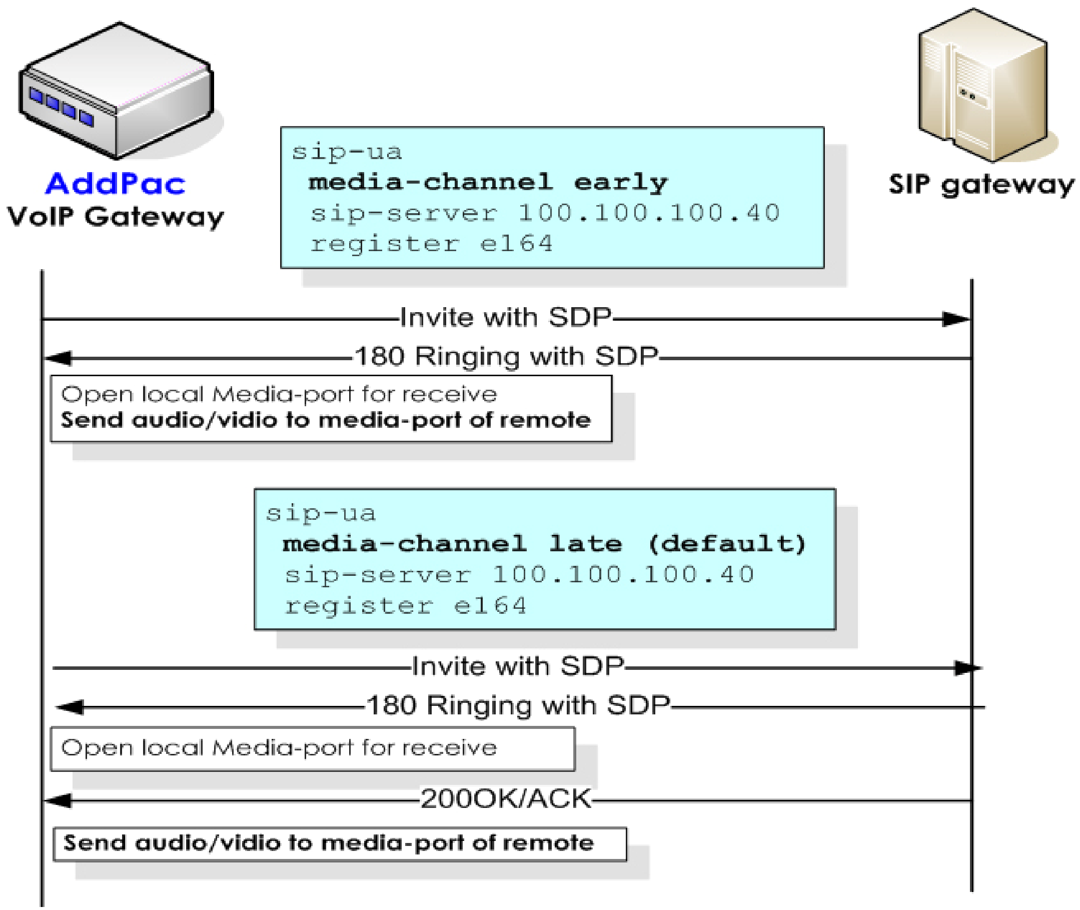
```
Sending SIP PDU to ( 100.1.1.40:5060 ) from 5060
REGISTER sip:100.1.1.40 SIP/2.0
Via:SIP/2.0/100.100.100.1:5060;branch=z9hG4bK5543002ea434
From: <sip:addpac1@100.1.1.40>;tag=5543002ea4
To: sip:addpac1@100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Contact: <sip: addpac1 @100.1.1.2>;expires=3600
Expires: 3600
Content-Length: 0
Max-Forwards: 70

Received SIP PDU from ( 100.100.100.40:5060 )
SIP/2.0 401 UNAUTHORIZED
To: sip: addpac1 @100.1.1.40;tag=10efb2232
From: <sip: addpac1 @100.1.1.40>;tag=5543002ea4
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea434
WWW-Authenticate: Digest realm="100.1.1.40",
nonce="33f16bf255ec473dd45911db0d0a9dc1", algorithm=MD5

Sending SIP PDU to ( 100.1.1.40:5060 ) from 5060
REGISTER sip:registrar.100.1.1.40 SIP/2.0
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea435
From: <sip: addpac1 @100.1.1.40>;tag=5543002ea4
To: sip: addpac1 @100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 35 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Authorization: Digest username="addpac1", realm="100.1.1.40",
nonce="33f16bf255ec473dd45911db0d0a9dc1", uri=".100.1.1.40", respon
se="8b72ee2edd93765a8b6ae8896fcb4136", algorithm=MD5
Contact: <sip: addpac1 @100.1.1.2>;expires=3600
Expires: 3600
Content-Length: 0
Max-Forwards: 70
```

```
Received SIP PDU from ( 100.100.100.40:5060 )
SIP/2.0 200 OK
P-Associated-Uri: <sip:8225683848@100.1.1.40>
To: sip: addpac1 @100.1.1.40;tag=2ae655e
From: <sip: addpac1 @100.1.1.40>;tag=5543002ea4
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 35 REGISTER

REGISTER sip:100.1.1.40 SIP/2.0
Via:SIP/2.0/100.100.100.1:5060;branch=z9hG4bK5543002ea434
From: <sip: addpac1 @100.1.1.40>;tag=5543002ea4
To: sip: addpac1 @100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Contact: <sip: addpac2 @61.33.161.71>;expires=3600
Expires: 3600
Content-Length: 0
Max-Forwards: 70

SIP/2.0 401 UNAUTHORIZED
To: sip: addpac2 @100.1.1.40;tag=10efb2232
From: <sip: addpac2 @100.1.1.40>;tag=5543002ea4
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 34 REGISTER
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea434
WWW-Authenticate: Digest realm="100.1.1.40",
nonce="33f16bf255ec473dd45911db0d0a9dc1", algorithm=MD5

Sending SIP PDU to ( 100.1.1.40:5060 ) from 5060
REGISTER sip:registrar.100.1.1.40 SIP/2.0
Via: SIP/2.0/UDP 61.33.161.71:5060;branch=z9hG4bK5543002ea435
From: <sip: addpac2 @100.1.1.40>;tag=5543002ea4
To: sip: addpac2 @100.1.1.40
Call-ID: 55416743-90fe-0077-802e-0002a4ffffea@61.33.161.71
CSeq: 35 REGISTER
Date: Tue, 01 Nov 2005 16:20:37 GMT
User-Agent: AddPac SIP Gateway
Authorization: Digest username="addpac2", realm="100.1.1.40",
```

VoIP Network Configuration

- SIP Proxy Related Configuration –

Media-channel early



VoIP Network Configuration

- SIP Proxy Related Configuration –

Timer Related command

Set

```
(config-sip-ua)# timeout ?
  texpires   set SIP INVITE request timeout value (sec)
  treg       set SIP REGISTER timeout value (sec)
  tregtry    set SIP REGISTER retry timeout value (sec)
  tretry     set SIP retry timeout value (msec)
  tsipping   set SIP PING timer
```

confirm

```
(config-sip-ua)# show sip
Proxyserver Registration Information
   proxyserver registration option = e164
   Proxyserver list :
      ---------------------------------------------------------------
         Server address            Port     Priority         Status
      ---------------------------------------------------------------
        100.1.1.40                 5060      128             Trying
   Proxyserver registration status :
      ----------------------------------------------
         UserName            Regist            Status
      ----------------------------------------------
         addpac1              yes            Trying
         addpac2              yes       Not Registered
SIP UA Timer counters
   retry counter = 10
SIP UA Timer values
   tretry (sip retry timer) = 500 msec.
   treg (sip register timer) = 60 sec.
   tregtry (sip register retry timer) = 20 sec.
   texpires (sip invite expire timer) = 180 sec.
   tsipping (sip ping timer) = 30 sec.
SIP UA MIN-SE value
   Min-SE = 1800 sec.
```

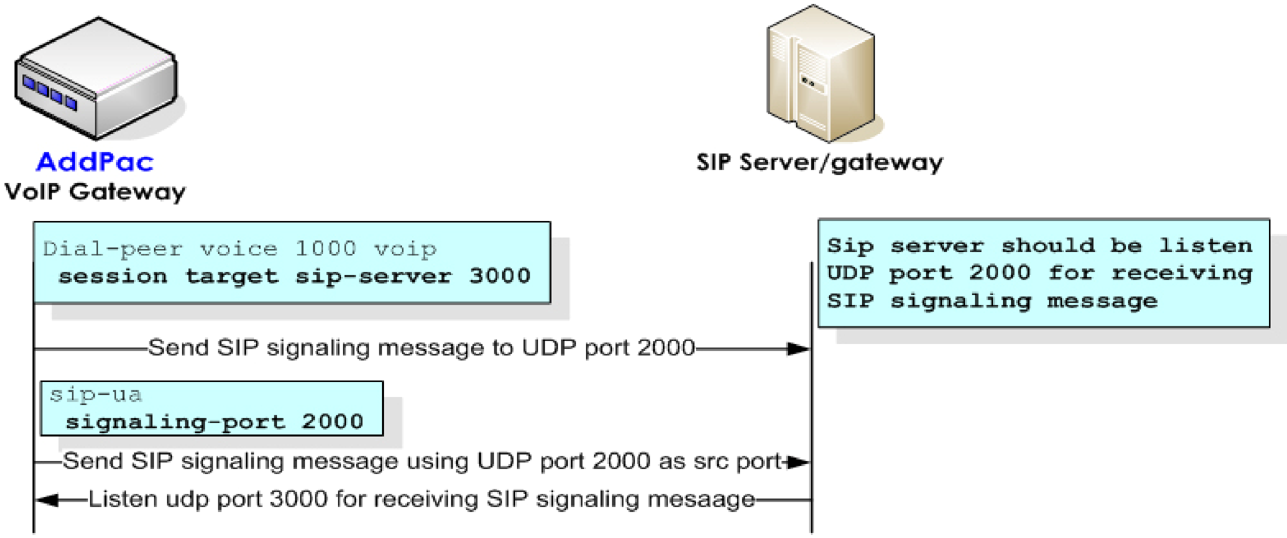
Registration Status

Activated Timer parameter

VoIP Network Configuration

- SIP Proxy Related Configuration –

Signaling-port/session target <sip-port>




```
Sending SIP PDU to ( 100.1.1.40:3000 ) from 2000
REGISTER sip:100.1.1.40 SIP/2.0
Via: SIP/2.0/UDP 100.1.1.2:2000;branch=z9hG4bK3e43f100a415
From: <sip:addpac1@100.1.1.40>;tag=3e43f100a4
To: sip:addpac1@100.1.1.40
Call-ID: 3efd6843-e91e-f167-8000-0002a4ffd0c0@61.33.161.72
CSeq: 15 REGISTER
Date: Wed, 02 Nov 2005 18:06:14 GMT
User-Agent: AddPac SIP Gateway
Contact: <sip:addpac1@100.1.1.2:2000>;expires=60
Expires: 60
Content-Length: 0
Max-Forwards: 70
```

Caution

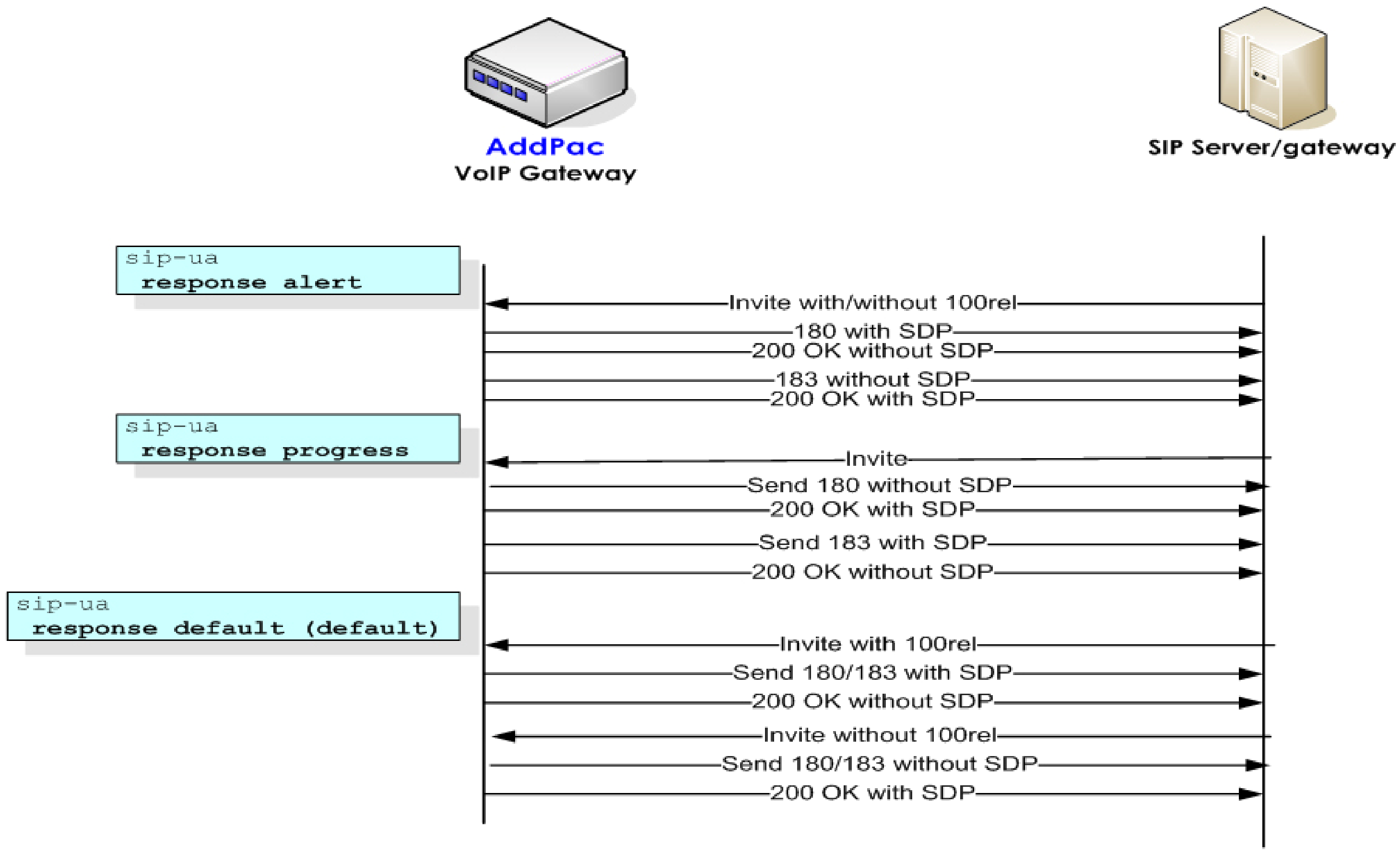
This command is needed "system reboot"

So configure then write and reboot.

VoIP Network Configuration

- SIP Proxy Related Configuration –

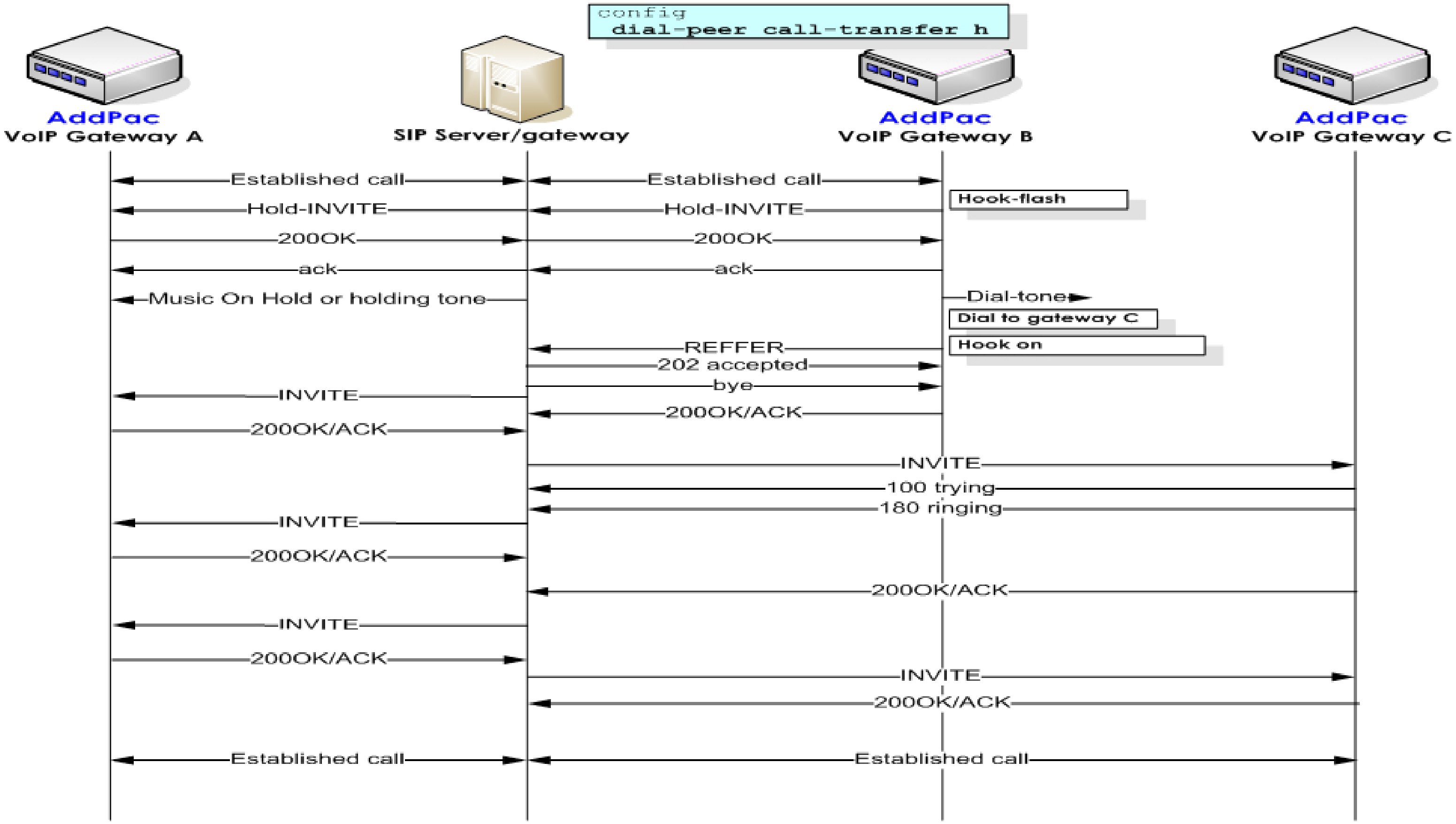
Response < alert | default | progress >



VoIP Network Configuration

- SIP Proxy Related Configuration –

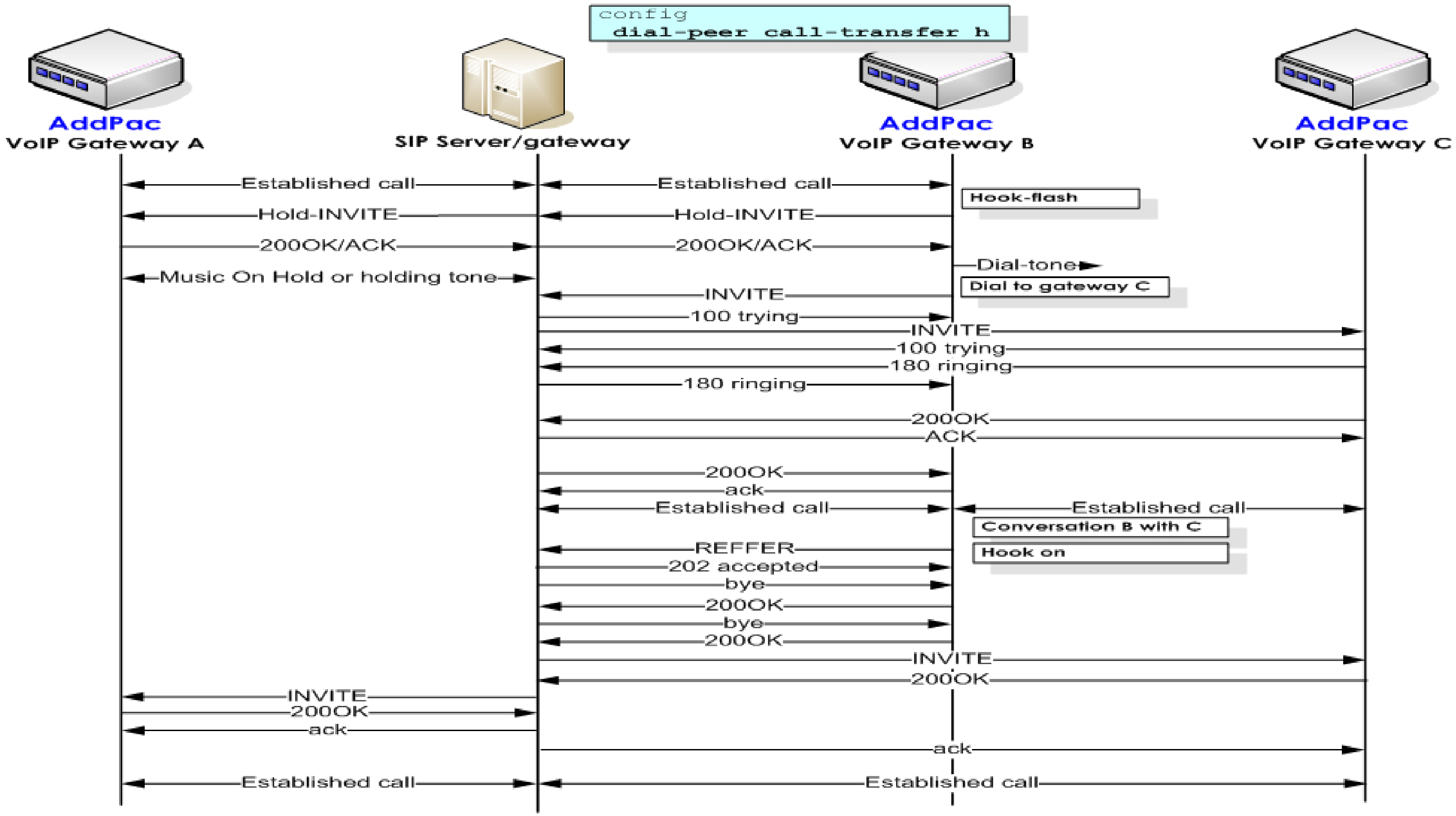
Blind Call Transfer by re-invite



VoIP Network Configuration

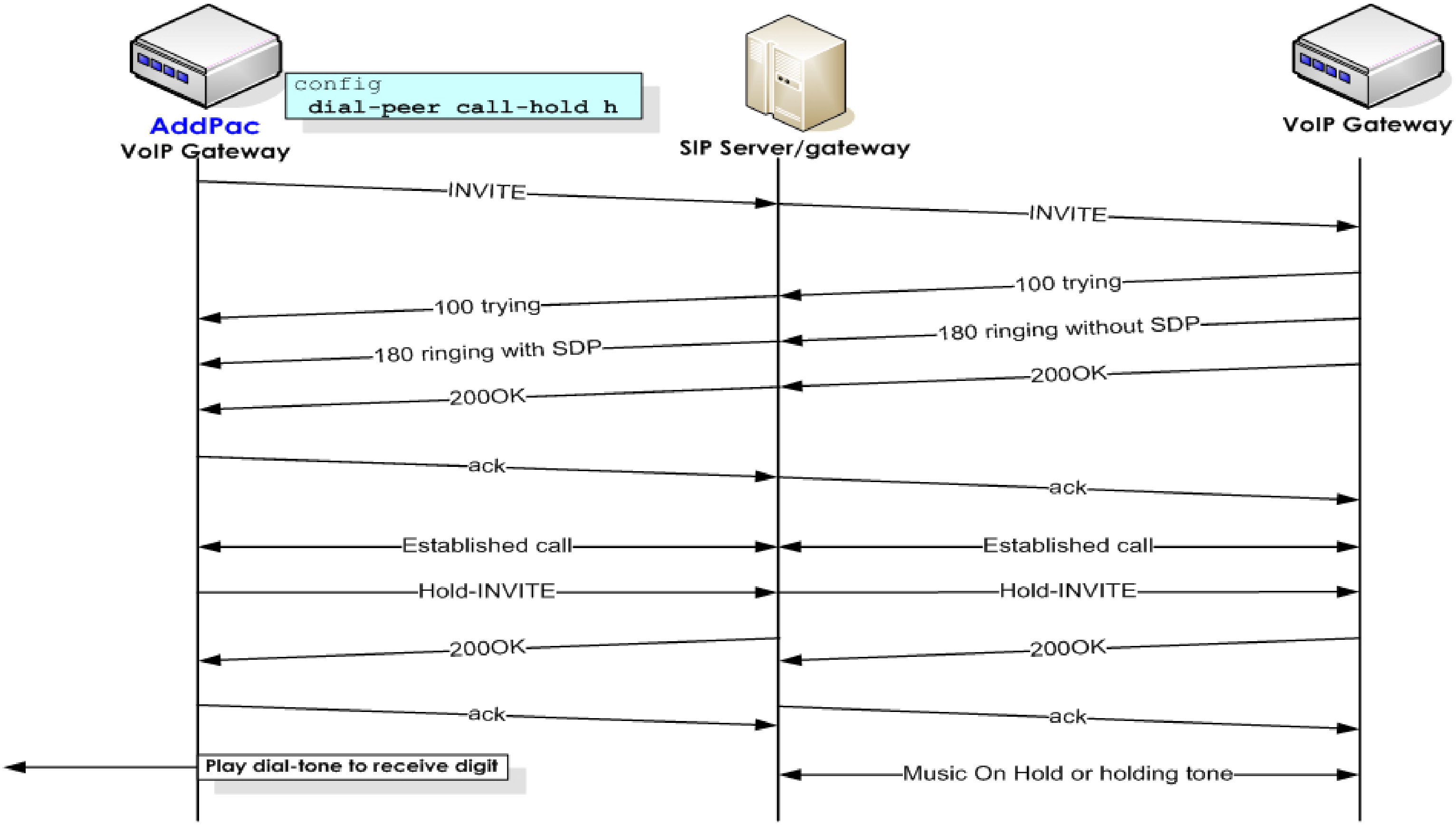
- SIP Proxy Related Configuration –

Attended Call Transfer by re-invite



VoIP Network Configuration
- SIP Proxy Related Configuration –

Call hold by re-invite



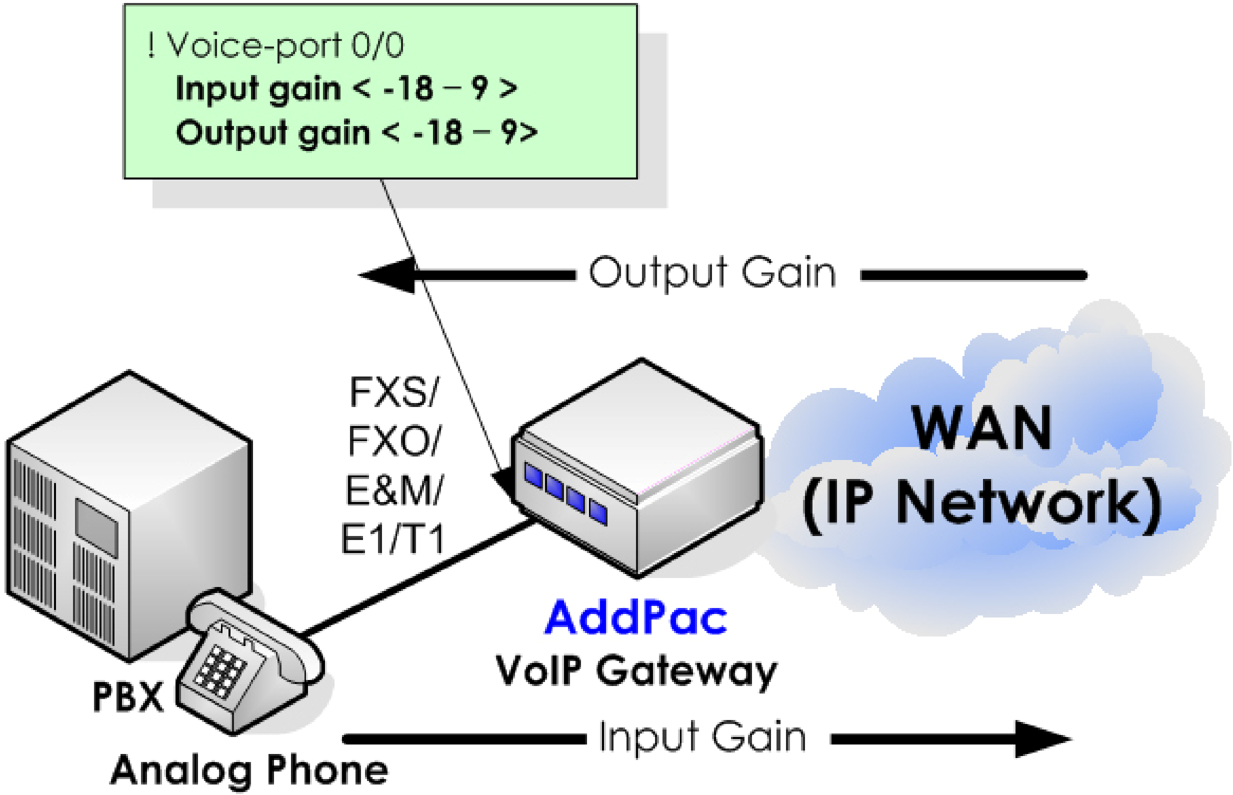
Voice Interface Configuration

Voice Interface Configuration

- Common
  gain
  CNG (Comfort Noise Generation)
  DTMF gain
  PLAR (Private Line Auto Ring-down)
- FXS/FXO
  Pin Assignment
  Ring Cadence/Frequency
  Polarity Inverse Detect/Generation
  Caller ID Detect/Generation
- E&M
  Pin Assignment E1/T1
- E1/T1
  Pin Assignment

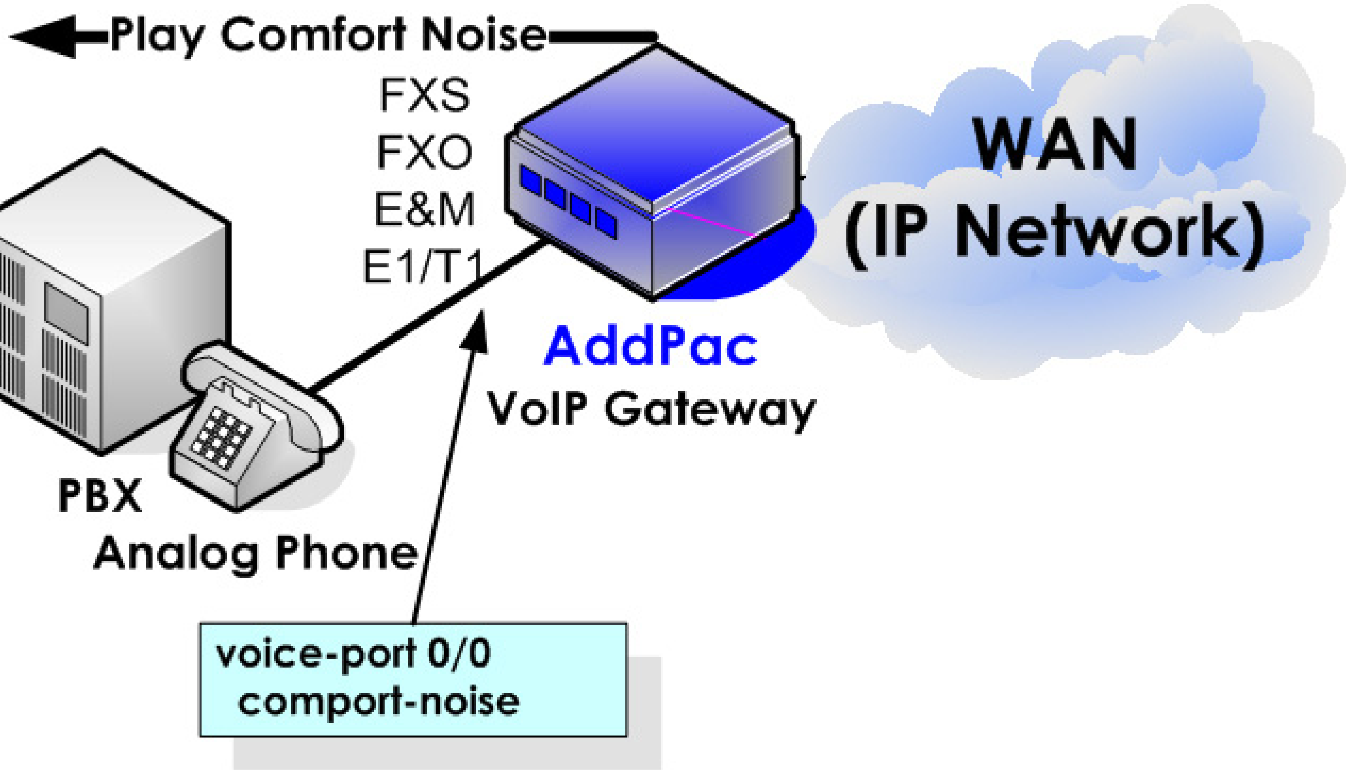
Voice Interface Configuration
- Common (1/4) –

Gain



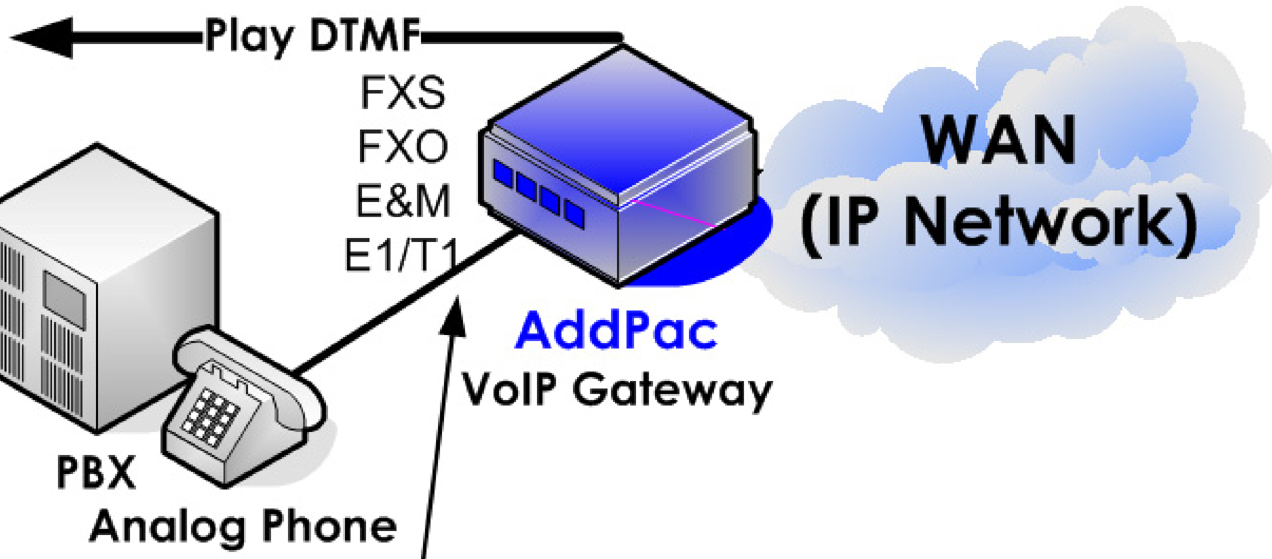
Voice Interface Configuration
- Common (2/4) –

Comfort Nose Generation



Voice Interface Configuration
- Common (3/4) –

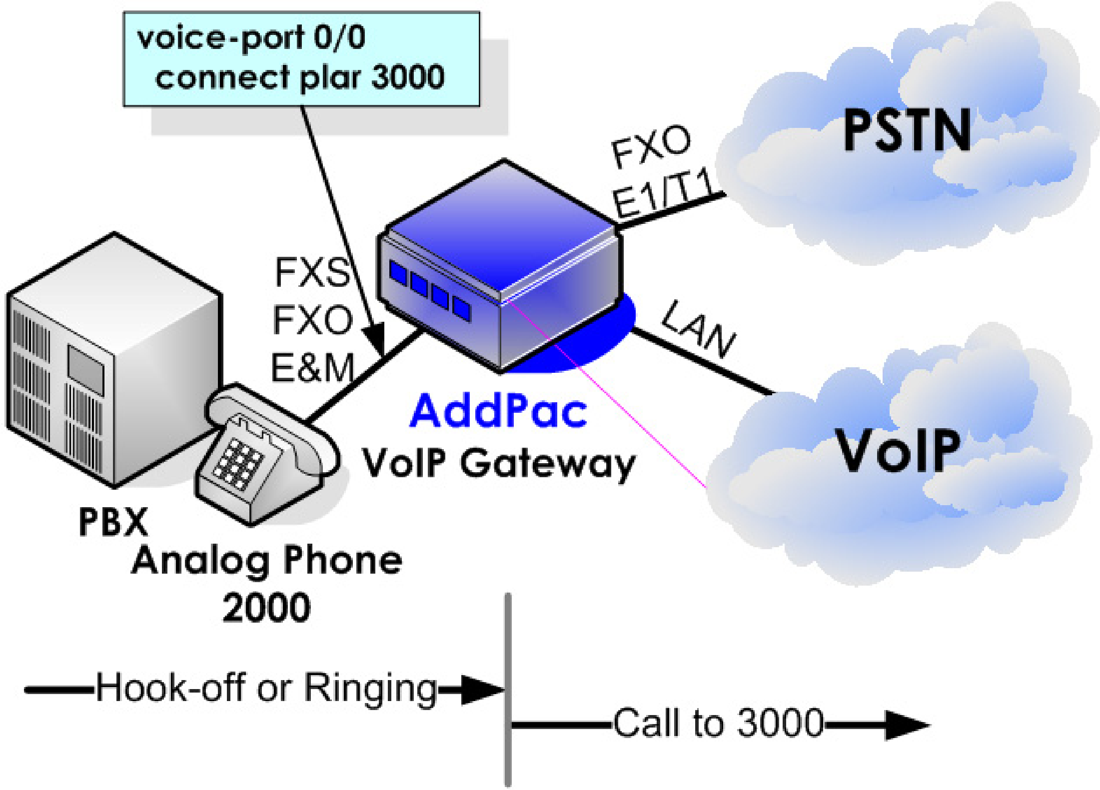
DTMF-gain




```
voice-port 0/0
  high-dtmf-gain <-31 ~ 3>
  low-dtmf-gain <-31 ~ 3>
```

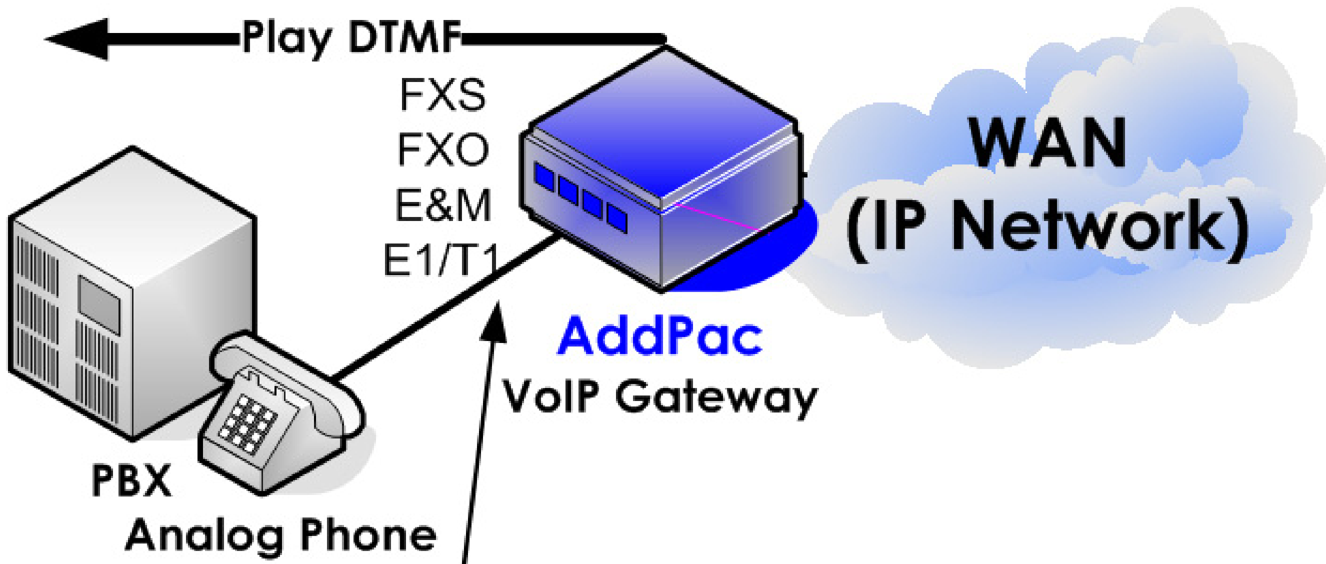
Voice Interface Configuration
- Common (4/4) –

PLAR(Private Line Auto Ring-down)



Voice Interface Configuration
- FXS/FXO (1/3) –

Ring cadence/frequency (FXS only)

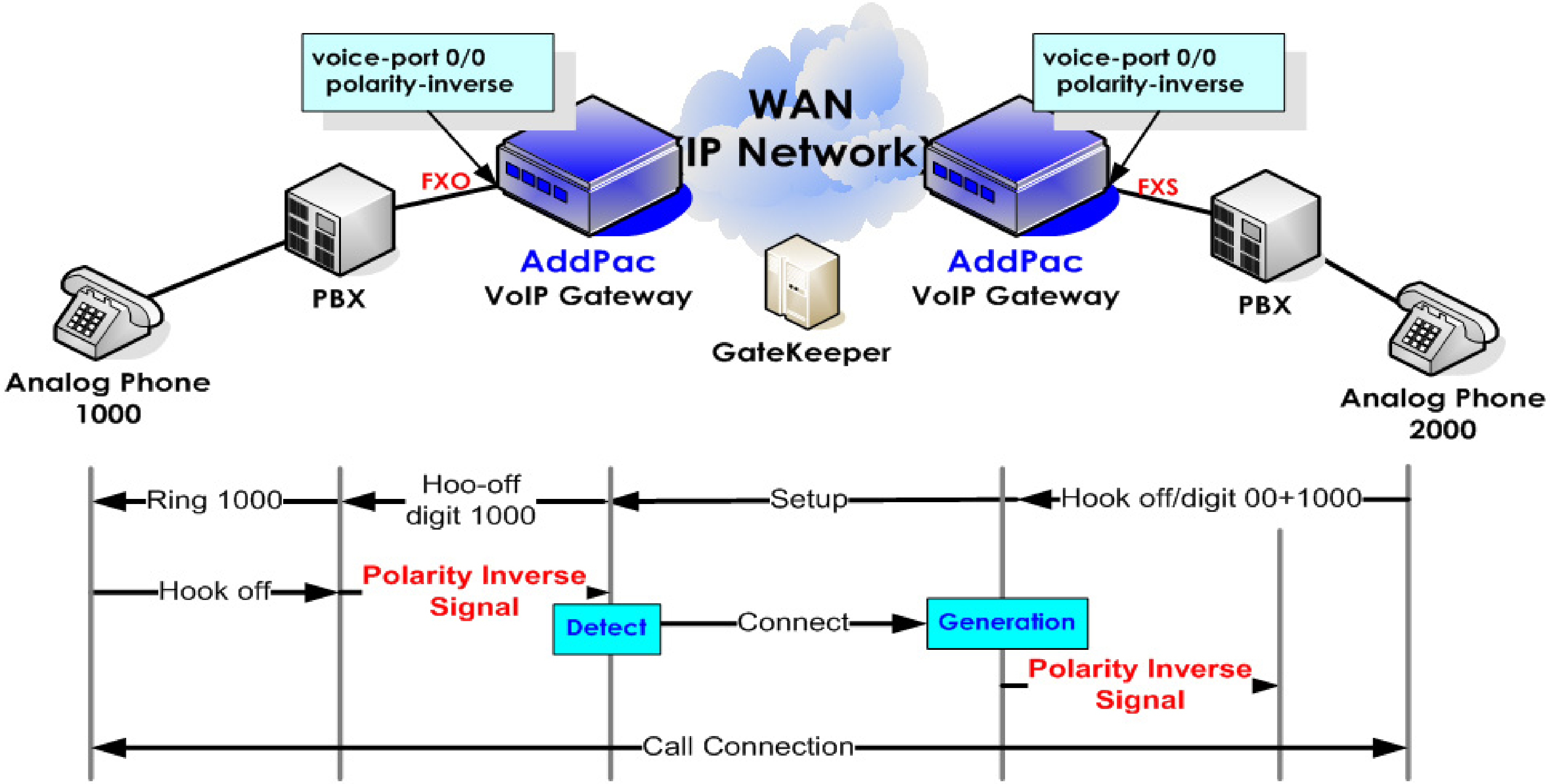



```
voice-port 0/0
  high-dtmf-gain <-31 ~ 3>
  low-dtmf-gain <-31 ~ 3>
```

Voice Interface Configuration

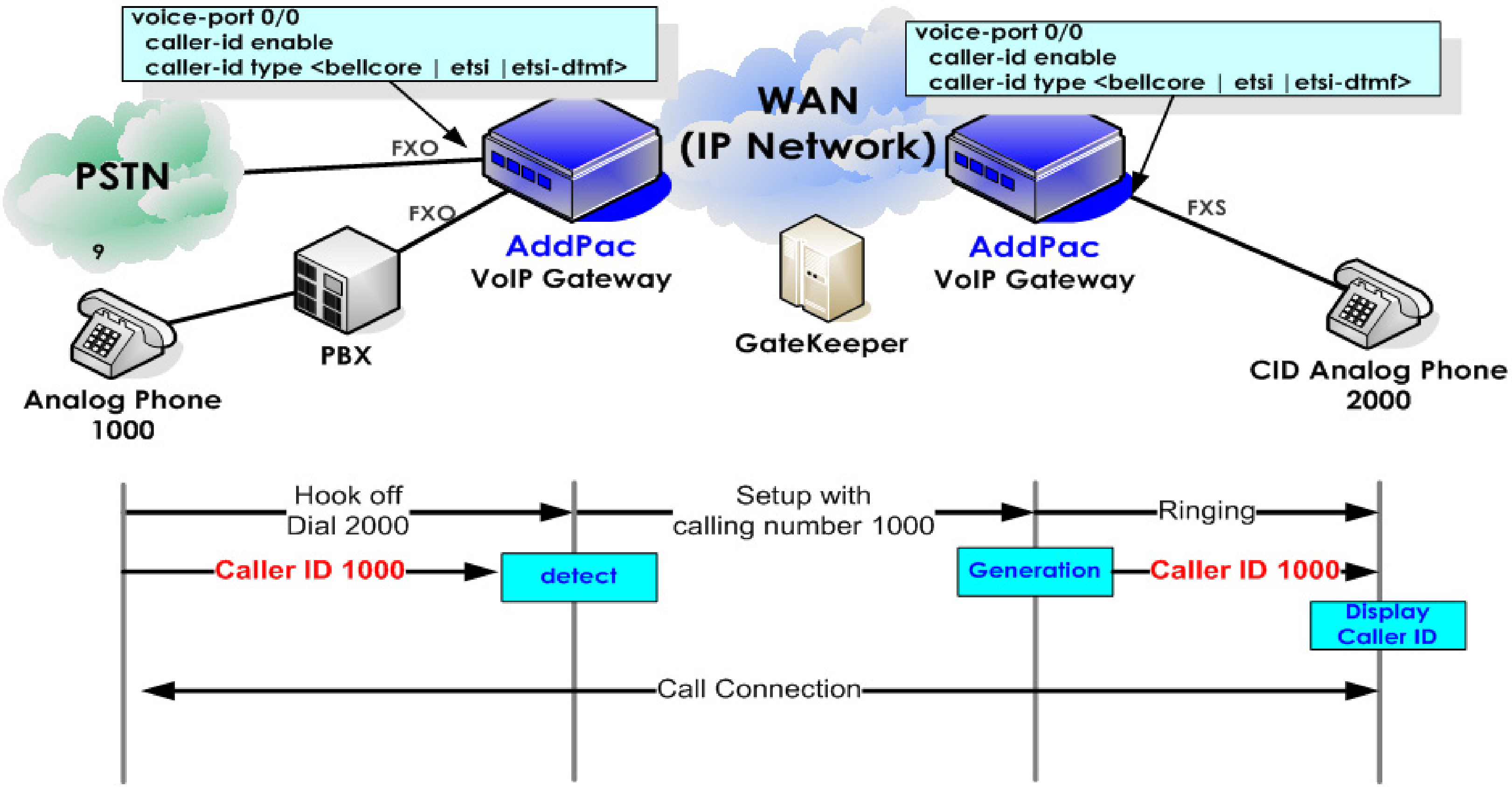
- FXS/FXO (2/3) –

Polarity Inverse detect (FXO) / Generation (FXS)



Voice Interface Configuration
- FXS/FXO (3/3) –

Caller ID detect (FXO) / Generation (FXS)



Voice Interface Configuration
- E1/T1 –

E1/T1 Related Command

ISDN-PRI

```
!
controller e1(t1) 0/0
signalling-type isdn
channel-group timeslots <1-31> <0-?>
isdn protocol-emulate <network|user>
Out-boddle

!
voice-port 0/0
! E1(t1)
  compand-type u-law
!
....
....
HO# show controller 0/0
Controller E1 slot(0)/port(0)
   T1 Link is UP
      No Alarm detected.
      Applique type is Channelized
.............
.........
```

R2/DTMF

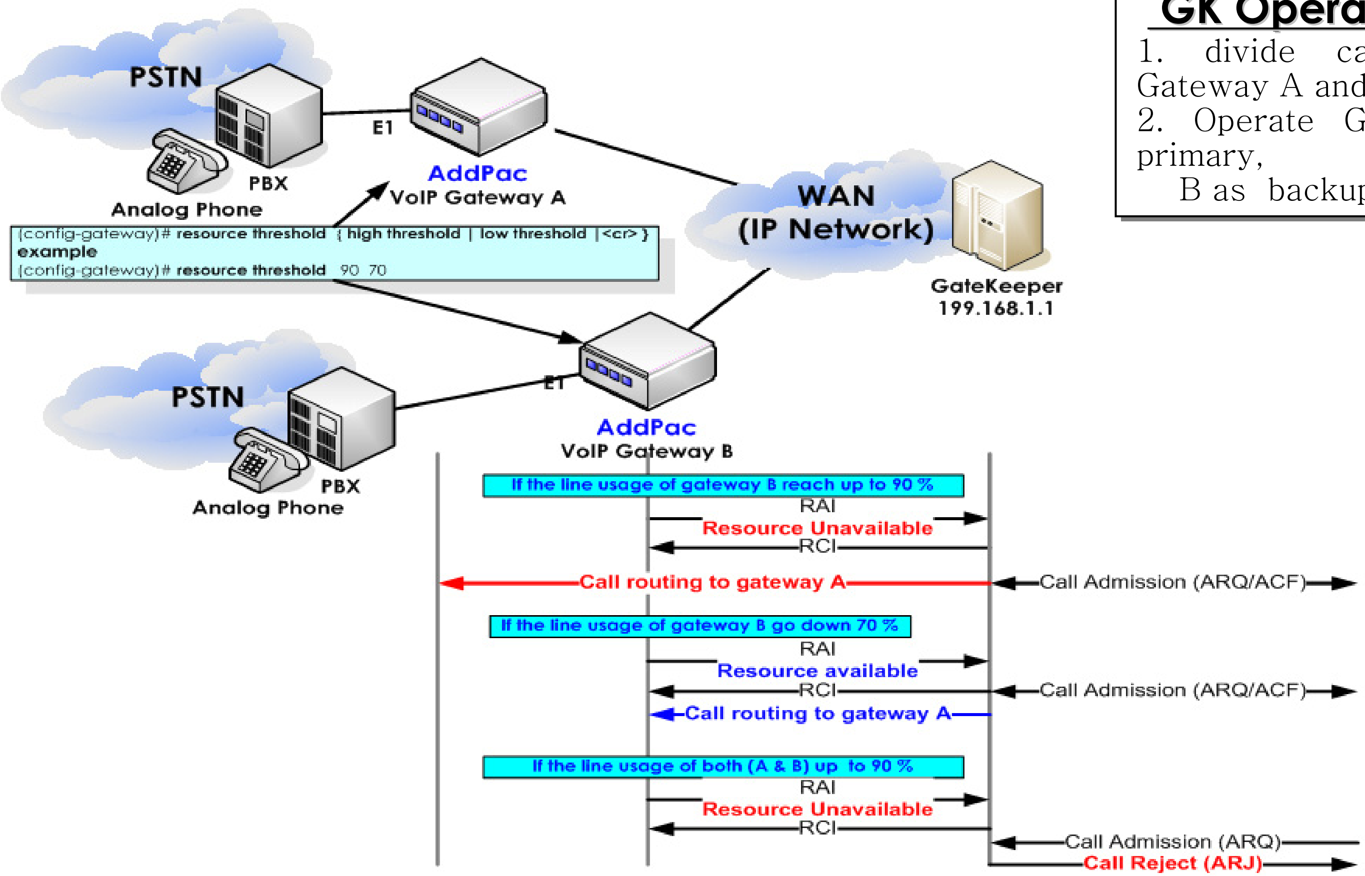
```
!
controller e1(t1) 0/0
signalling-type <r2 | dtmf>
channel-group timeslots <1-31> <0-?>
R2 get callid <R2 MFC only>
Out-boddle

!
voice-port 0/0
! E1(t1)
  compand-type u-law
!
....
....
HO# show controller 0/0
Controller T1 slot(0)/port(0)
   T1 Link is UP
      No Alarm detected.
      Applique type is Channelized
.............
.........
```

VoIP gateway Tuning

- Trunk Gateway Redundancy (RAI)
- Policy for poor VoIP network
- Number Plan
- PSTN Backup by busyout
- number Translation
- The solution for FXO Port block

Basic and Advanced Configuration
- Trunk Gateway Redundancy Policy(1/2) –

Through GK

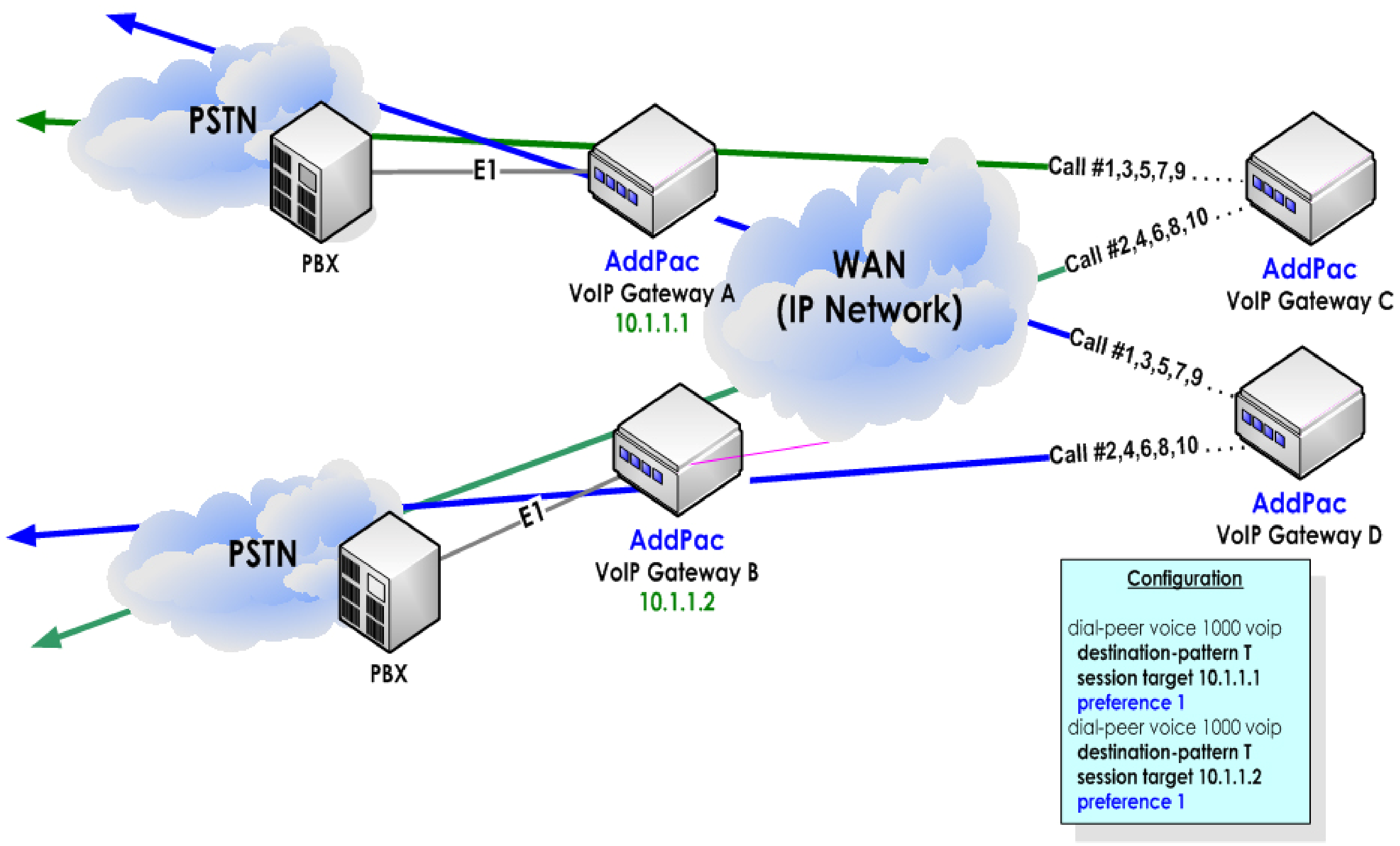


GK Operation Policy

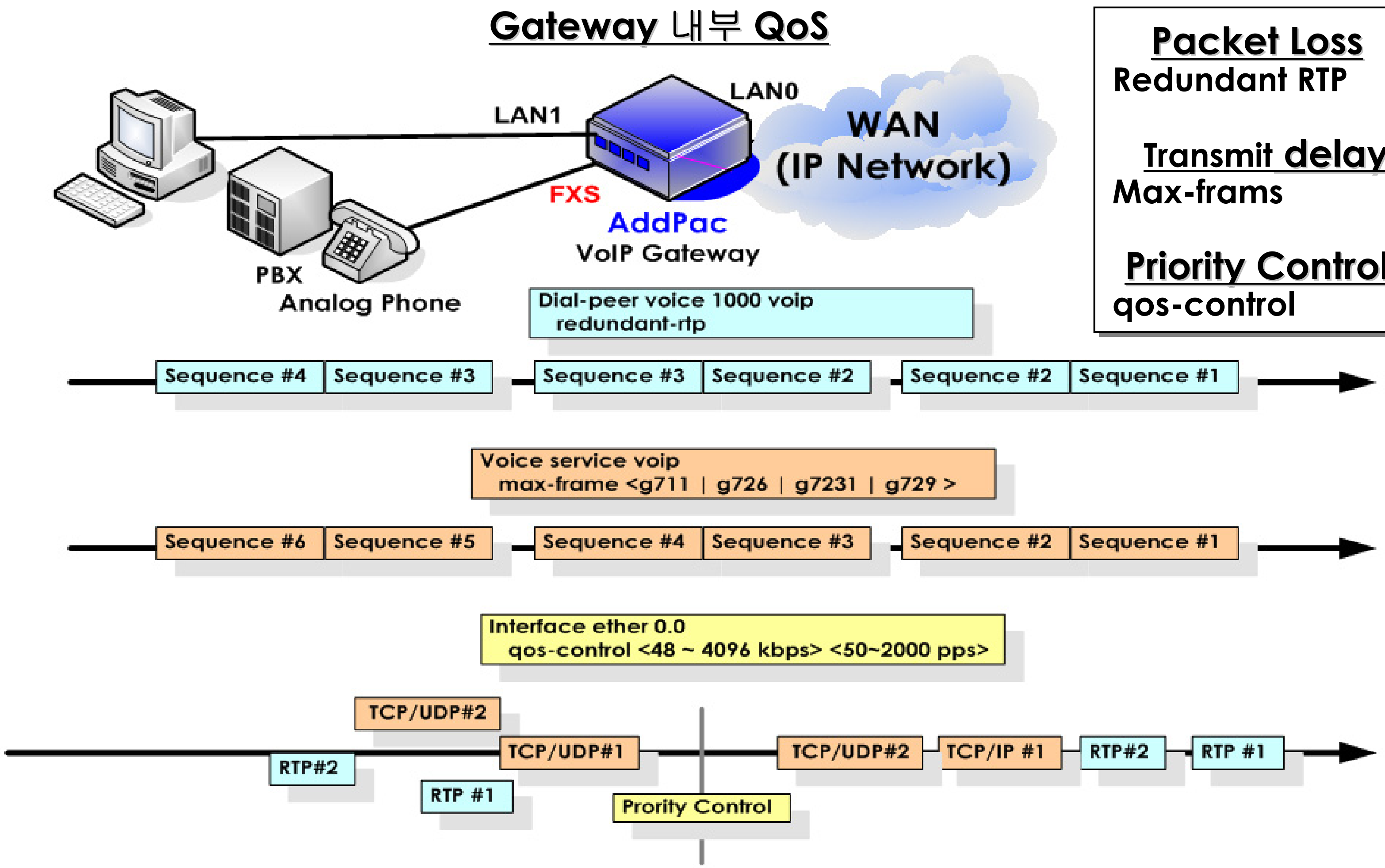
1. divide call traffic to Gateway A and B with 50:50
2. Operate Gateway A as primary,
   B as backup

Basic and Advanced Configuration
- Trunk Gateway Redundancy Policy(2/2) –

Traffic dispersion



Basic and Advanced Configuration
- Policy for poor VoIP Network (1/2) –



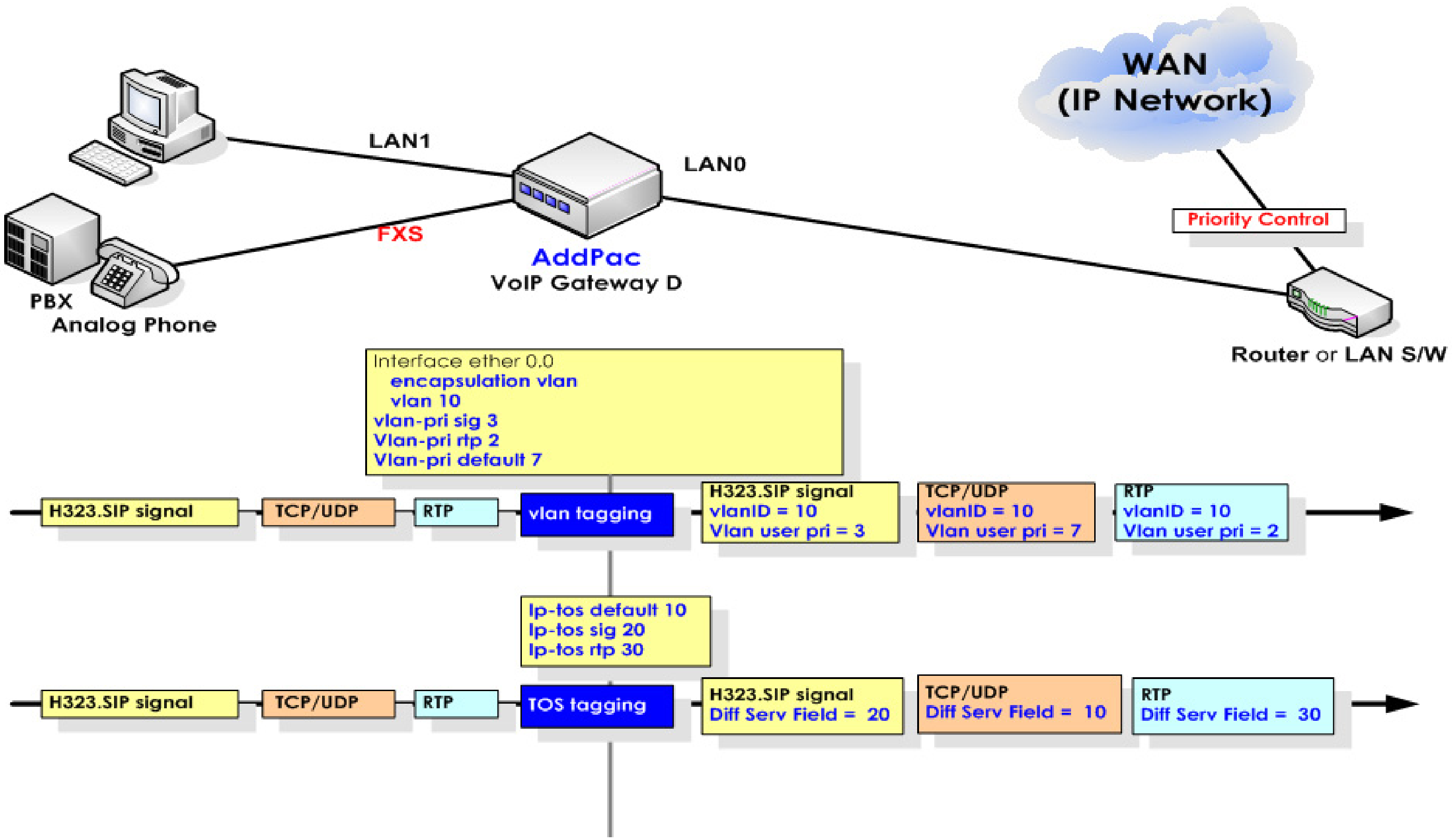
Packet Loss
Redundant RTP

Transmit delay
Max-frams

Priority Control
qos-control

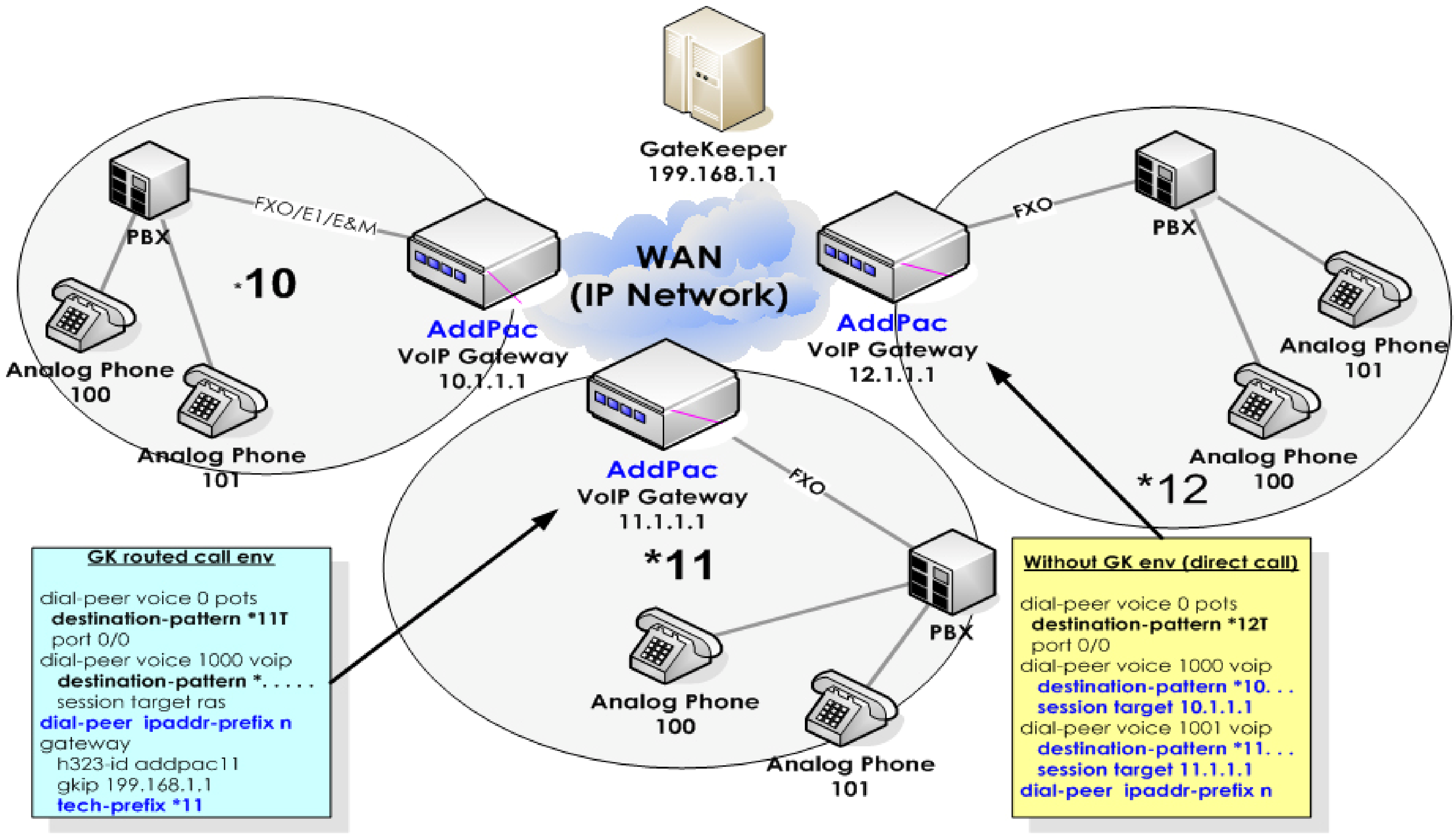
Basic and Advanced Configuration
- Policy for poor VoIP Network (2/2) –

VLAN/IP-TOS Field Tagging



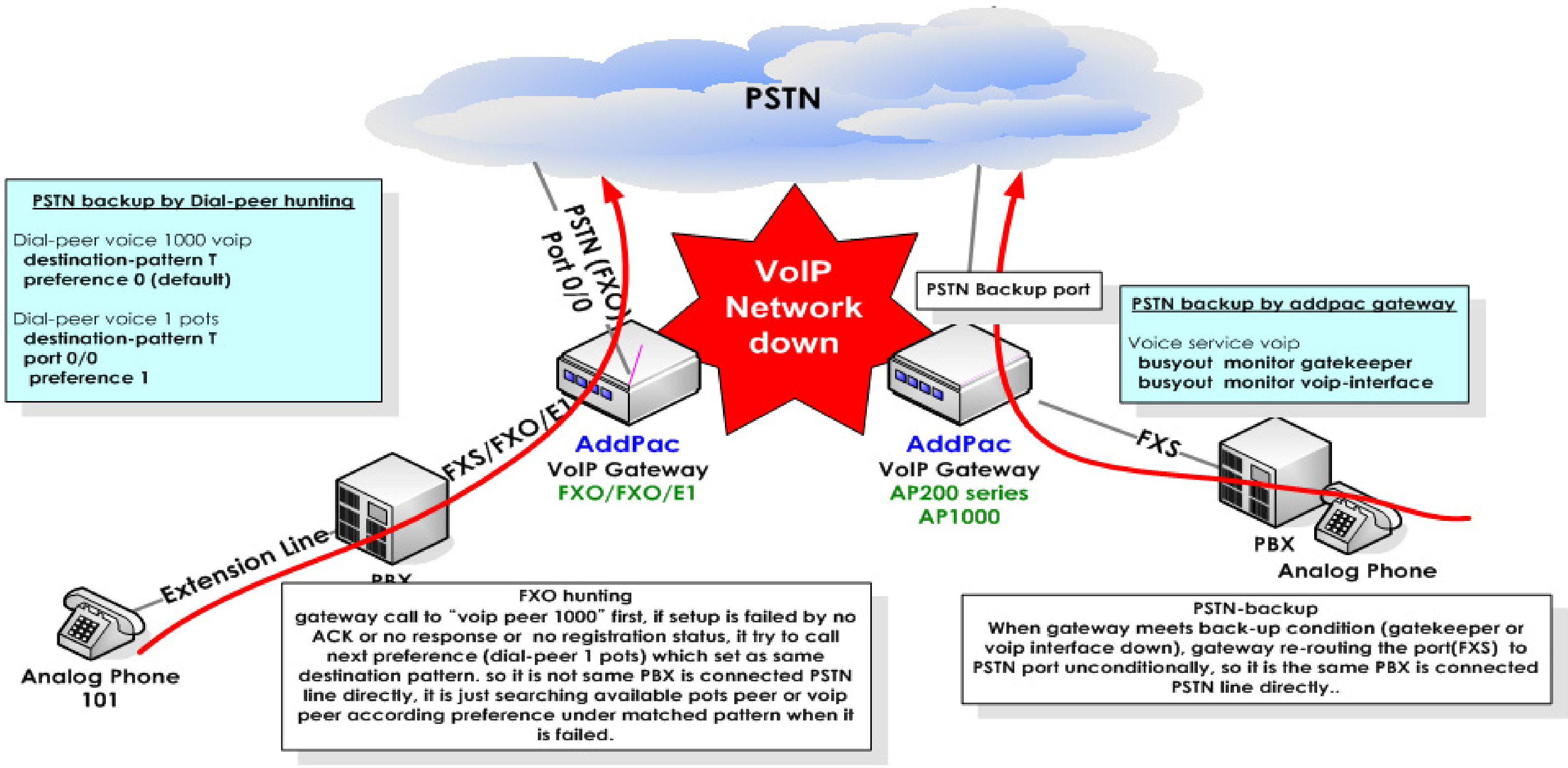
Basic and Advanced Configuration
- Number Plan (1/2) –

Private VoIP telepony Network



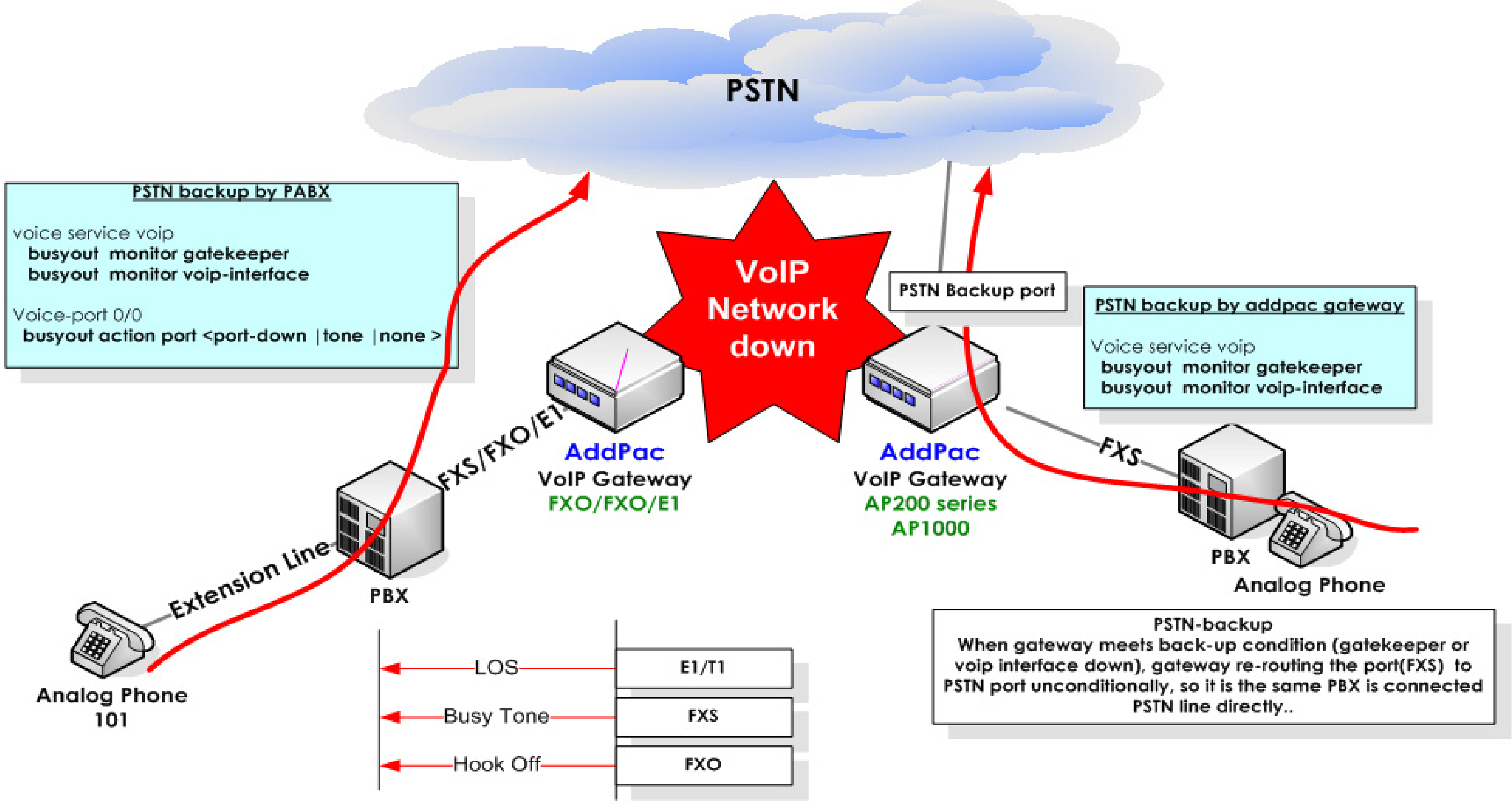
Basic and Advanced Configuration
- Number Plan (2/2) –

Call Re-routing by Dial-peer Hunting



The Difference between PSTN Backup and FXO hunting

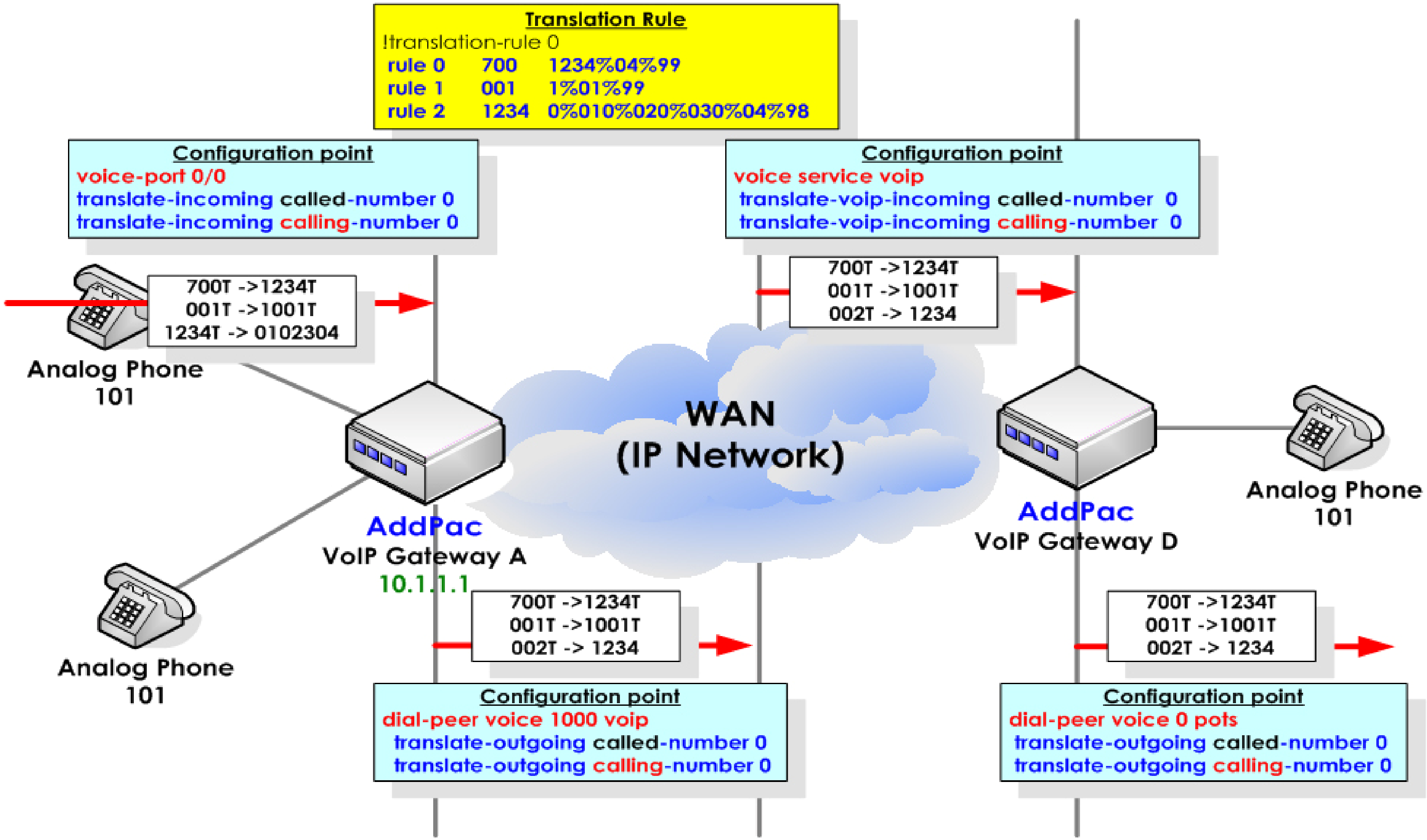
Basic and Advanced Configuration
- PSTN backup by busyout –



PSTN Backup by Busyout Action through PABX

Basic and Advanced Configuration

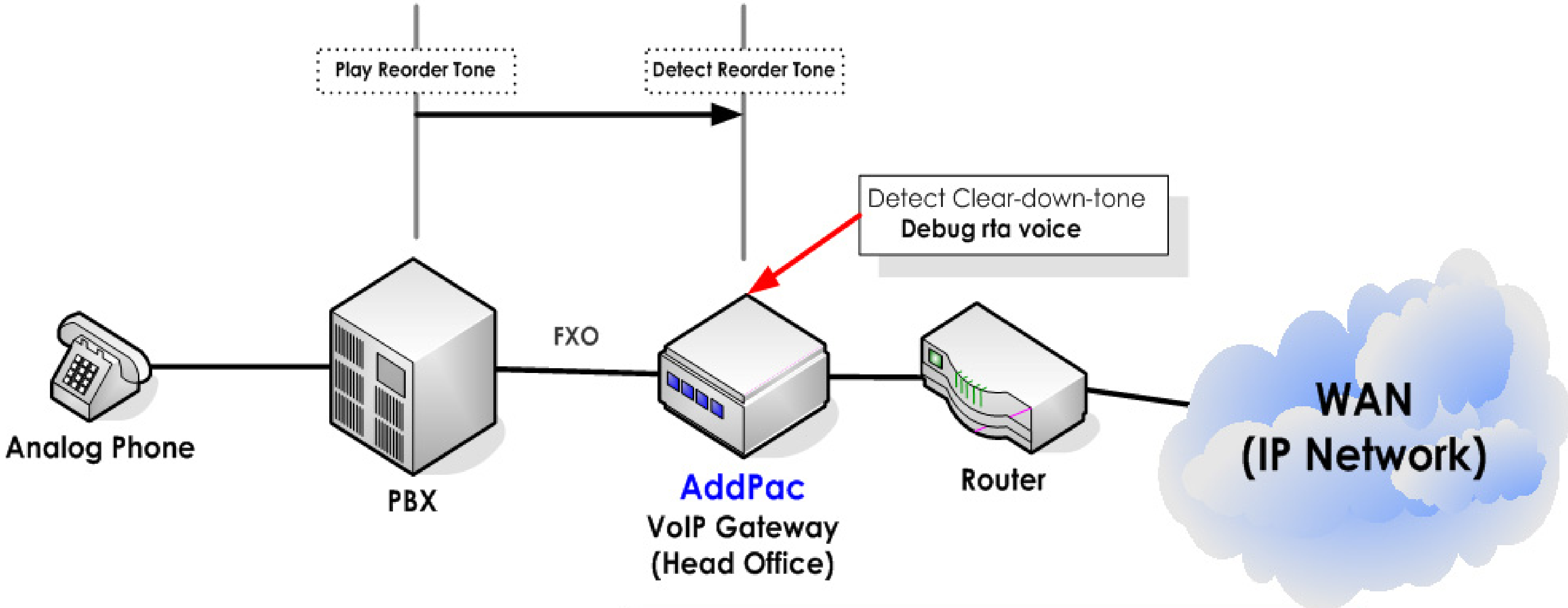
- number Translation –



Basic and Advanced Configuration

- The Solution for FXO port block (1/4) –

Clear Call by Clear down-tone frequency detect



Set value
voice class reorder-tone
<low preq> <high-preq> <on-time> <off-time>

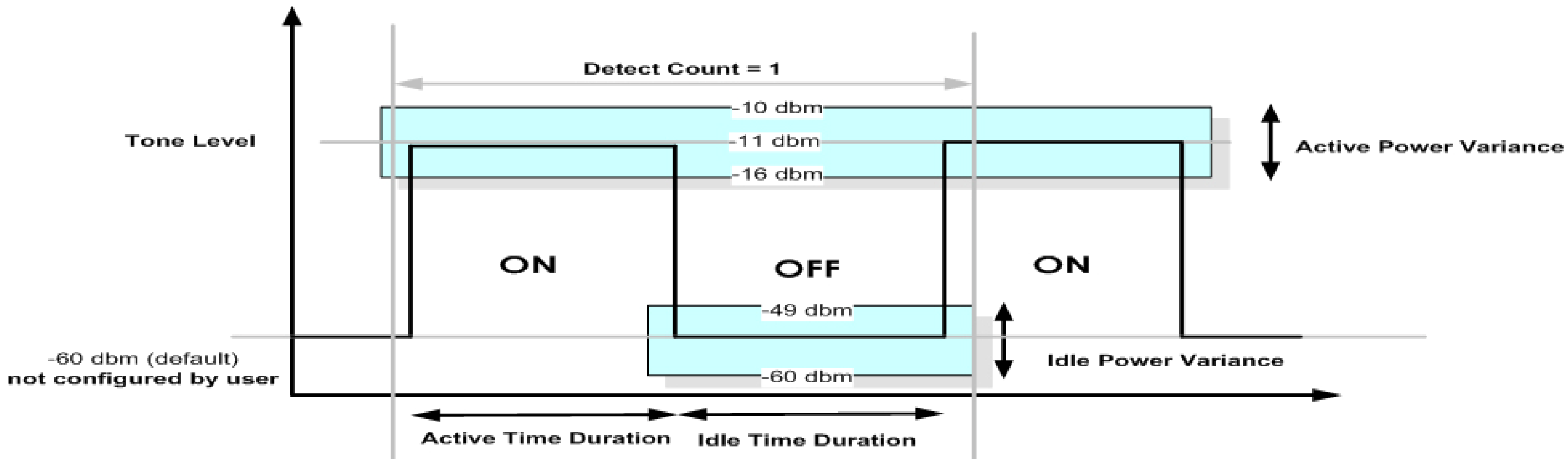
Clear Call by clear down-tone energy level detect(1/2)

(Example (In case of codec = g7231))

```
Codec=G711,G729 (10msec * count), G7231 (30msec * count )

# debug rta voice
Make PABX play clear-down-tone by hook-on the phone which connected extension line.
8 18 18 17 17 17 17 17 17 18 18 18 18 18 18 14 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11
11 16 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55 56 55 57 55 57 55 57 55 15 10 12 11 11 12 10 12 11 11 12
10 12 11 11 12 10 12 11 11 12 10 12 11 11 16 50 57 55 57 57 57 56 57 56 57 57 57 55 56 56 56 54 57 55 58 56 57 55 58 15 10 12 11 11
12 10 12 11 11 12 10 12 11 11 12 11 12 11 11 12 10 12 11 11 16 50 56 55 56 56 57 56 57 56 57 56 58 56 57 56 57 54 57 55 57 56 57 54
58 15 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11
```

Calculation

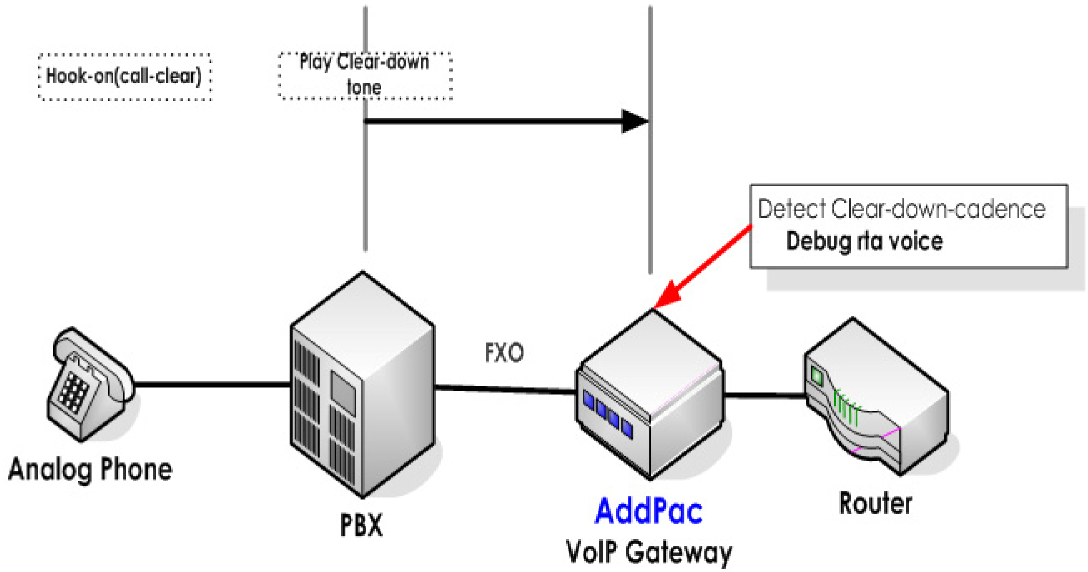
count of Tone Level (-10 ~ -16dbm) = 25
Active Time Duration (25 * 30ms) = 750ms, if codec is g711 or g729, it will be codec 250ms (25 * 10ms)
count of Idle level (-49 ~ -60dbm) = 25
Idle Time Duration (25 * 30ms) = 750ms, if codec is g711 or g729, it will be codec 250ms (25 * 10ms)
Tone Level = -11
Active Power Variance = 5 (-6 ~ 16dbm)
Idle Power Variance = 11 (-49 ~ -60)
Idle Tone Level is set as -60 dbm internally, so it was calculated as (11 = (-60(Min)) - (49(Max))).



Basic and Advanced Configuration

- The Solution for FXO port block (3/4) –

Clear Call by clear down-tone energy level detect(2/2)



Set the value which is detected by debugging

config
voice class clear-down-cadence <detect-counter> <tone-level> <active-time-duration> <idle-time-duration> <active-power-duration> <idle-power-duration>

Basic and Advanced Configuration

- The Solution for FXO port block (4/4) –

Clear Call by silence Tone detect

(Example (In case of codec = g7231))

Codec=G711,G729 (10msec * count), G7231 (30msec * count)

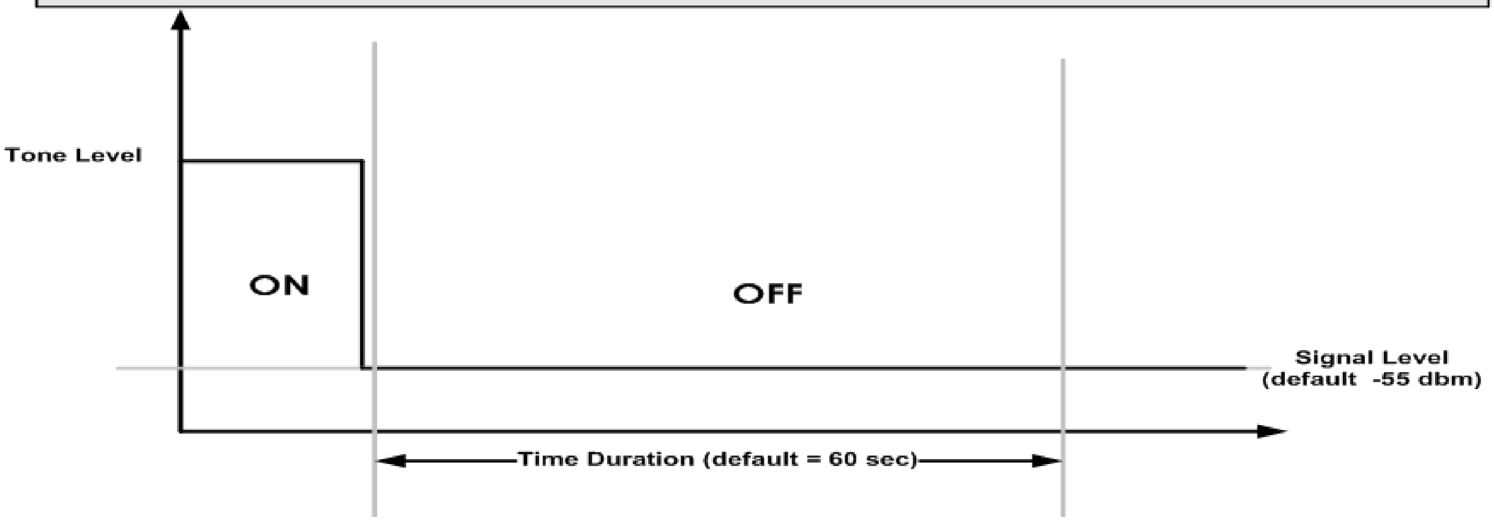
```
# debug rta voice
Make PABX play clear-down-tone by hook-on the phone which connected extension line.
18 18 18 17 17 17 17 17 17 18 18 18 18 18 18 14 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55 56 55 57 55 57 55
57 55 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55 56 55 57 55 57 55 57 55 49 58 56 57 54 58 56 53 56
57 56 56 55 57 56 57 55 56 55 57 55 57 55 57 55 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55 56 55 57
55 57 55 57 55 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55 56 55 57 55 57 55 57 55 49 58 56 57 54 58
56 53 56 57 56 56 55 57 56 57 55 56 55 57 55 57 55 57 55 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55
56 55 57 55 57 55 57 55
```

Calculation

Tone Level = -14 ~ -18 dbm

Time Duration = 30 sec

Signal Level = - 49 dbm



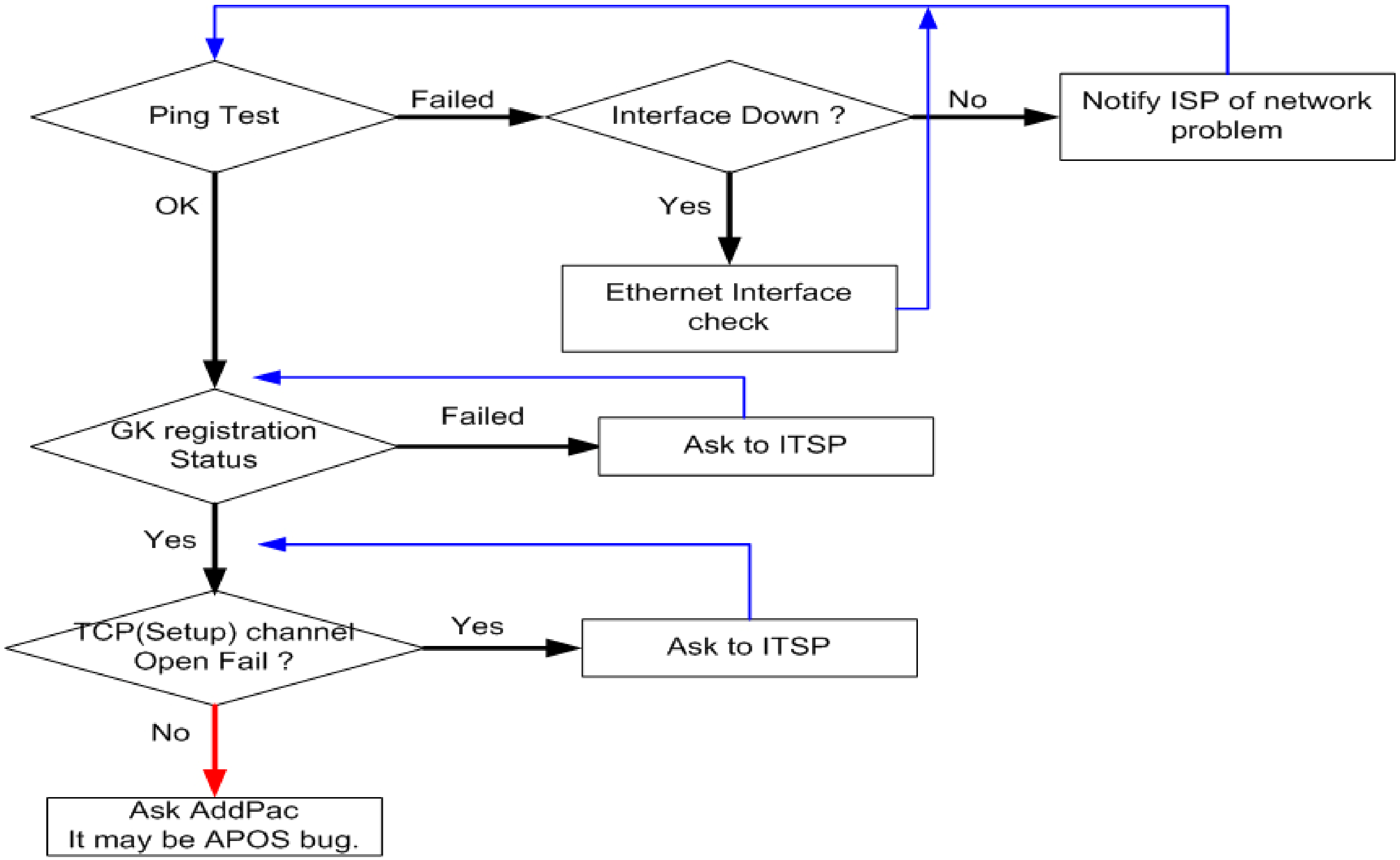
Chapter 3 Trouble Shooting

Trouble Shooting

- Trace Flow
- Debugging Command
- Call Trace

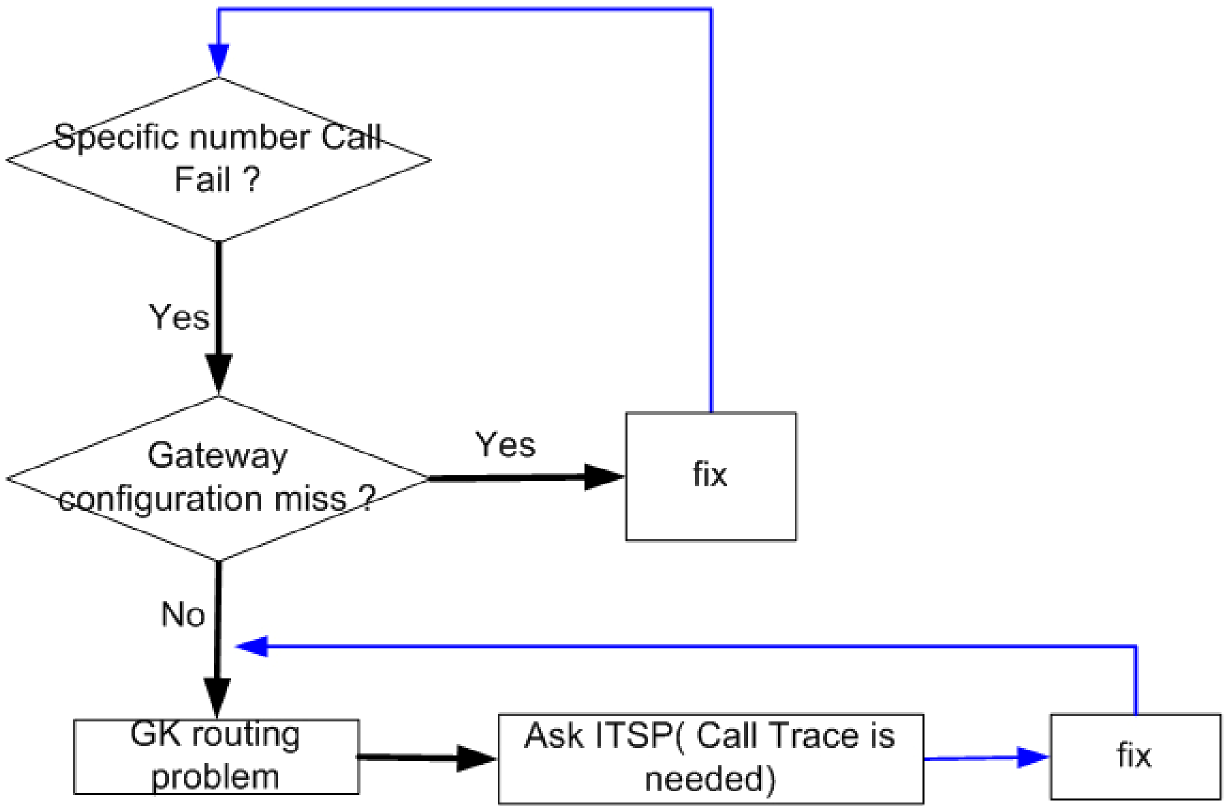
Trouble Shooting
- VoIP Network or GK (1/3) –

<u>FXO or PSTN backup after playing announcement about Internet fail or signal</u>



Trouble Shooting
- VoIP Network or GK (2/3) –

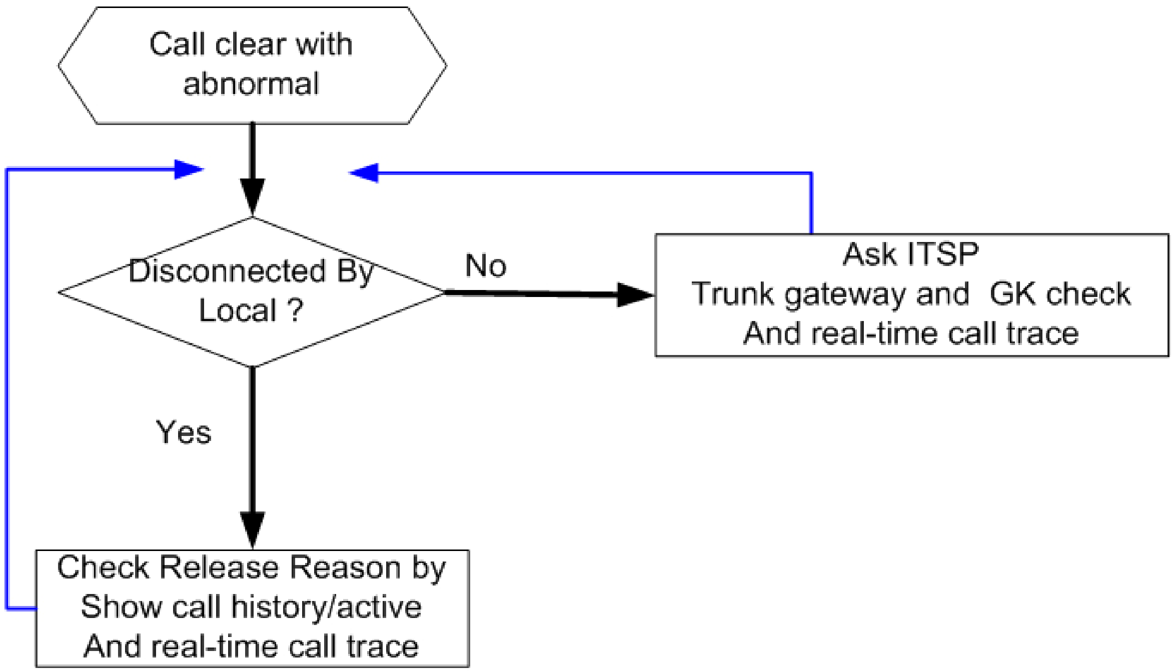
Specific number call fail



Trouble Shooting

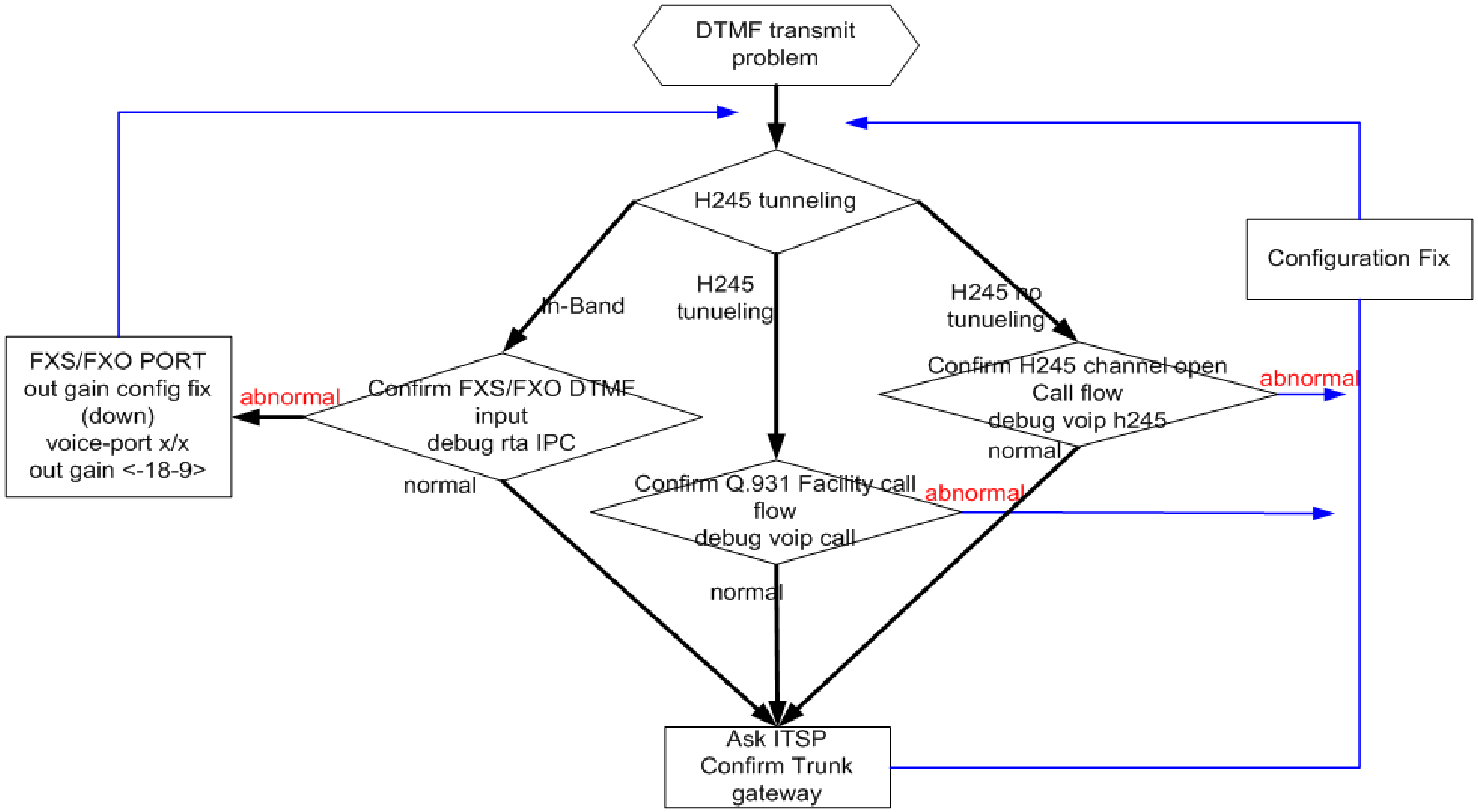
- VoIP Network or GK (3/3) –

Call Setup miss-operation or abnormal call clear



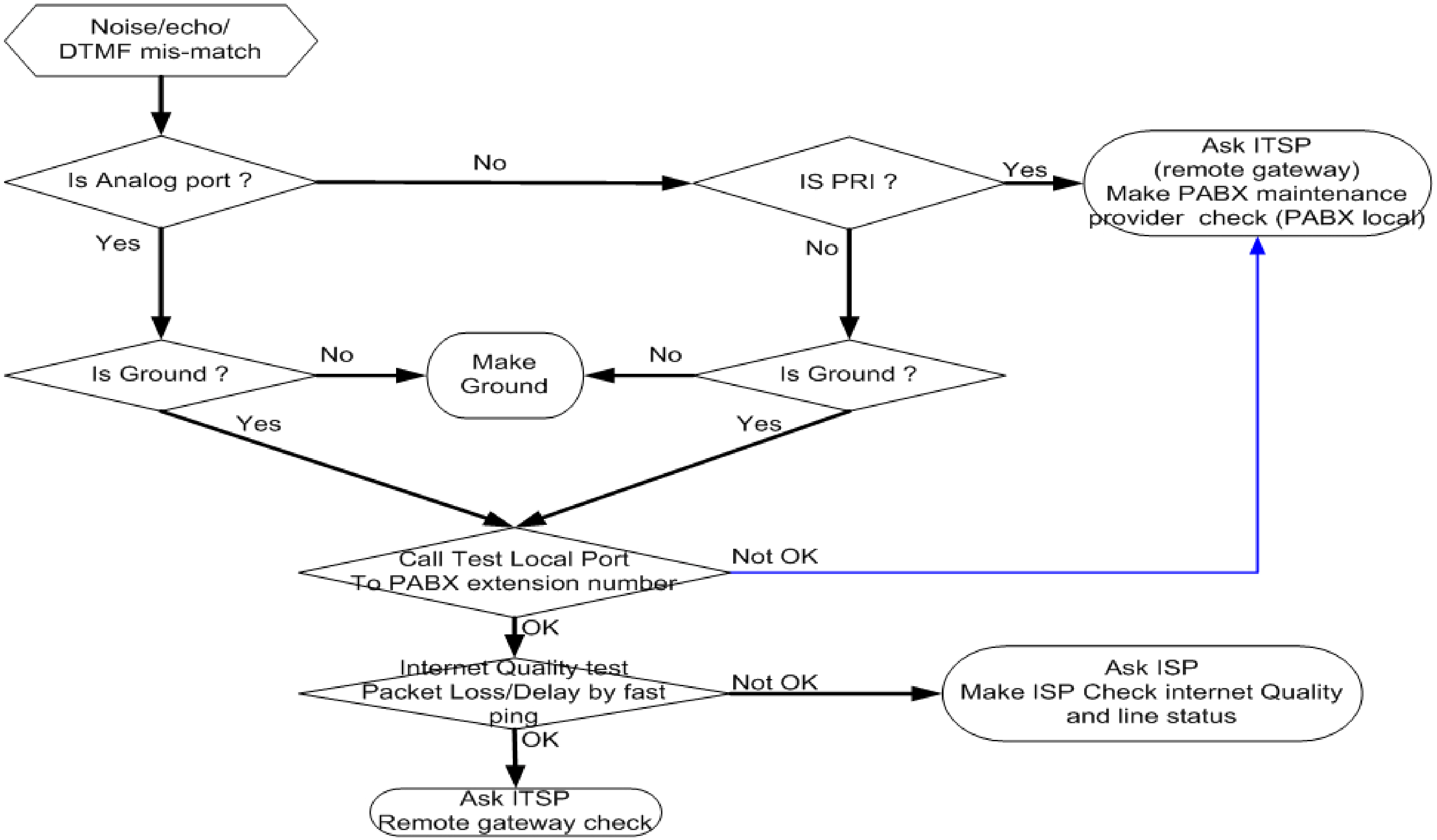
Trouble Shooting

- DTMF Relay or detect –

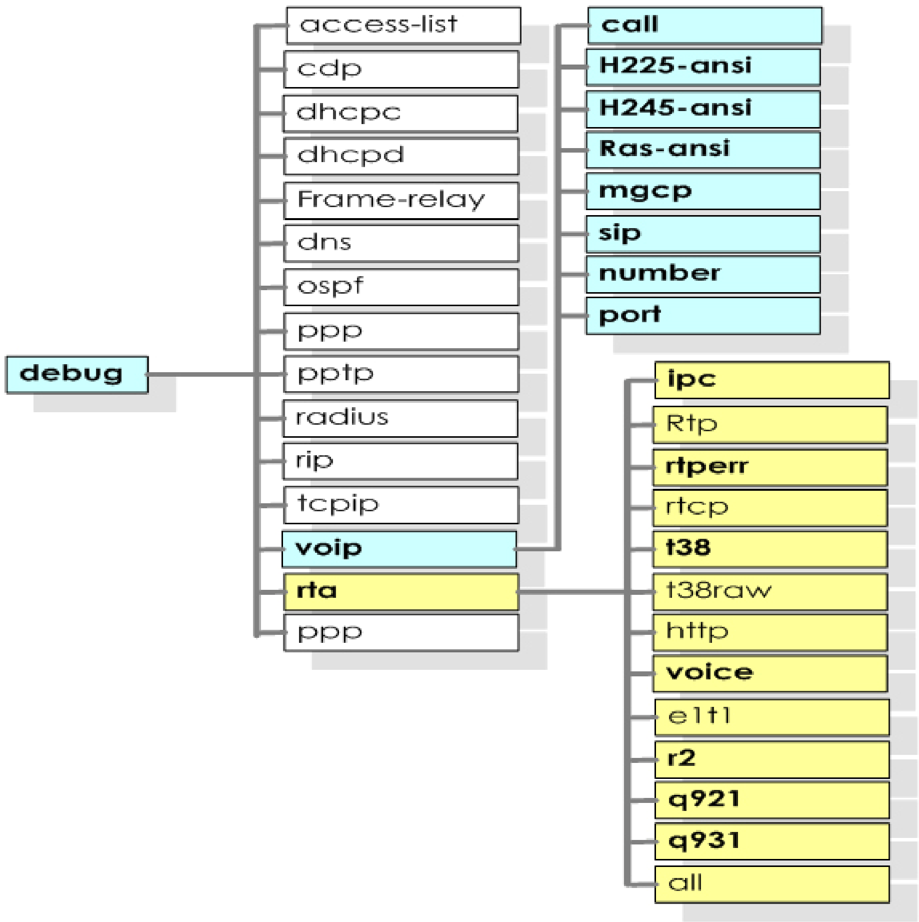


Trouble Shooting

- Voice Quality –



Debugging Command



Message display control

config)# debug-port

Usage

AP200# **debug voip call**
AP200# **config**
Enter configuration commands, one per line. End with CNTL/Z
AP200(config)# **debug-port**
AP200(config)# 4 <CEP 000000> : Call Received
5 <CEP 000000> : Call Initiated : calledNumber() callerInfo() crv(0) total(0)
6 <Call 286> :
****************** Call Created status(InitiatedByFXS)
****
*
7 <CEP 000000> : Call id(cc47e640-6a2b-67ca-8183-0002a400380d) callNum(286)
8 <Call 286>

call Trace

- RAS brief/detail (1/2)–

```
AP200#
AP200# debug voip call
AP200(config-gateway)# register
6       <GK     0>      : Try registration to All candidated GKs exclude() exclusi
           ve()
7       <Gk     0>      : Try registration to gk host(172.19.1.248) port(1719)
8       <GK     0>      : Send RRQ.
AP200(config-gateway)# 9       <GK     0>      : Received RCF from 'addpa
           c-private-gk1'.
10          <GK     0>      : Success registration to GK host(172.19.1.248) port
           (1719)
```

```
AP200# debug voip ras
AP200(config-gateway)# regis
<> PDU Encoding ...
value RasMessage ::= registrationRequest :
{
 requestSeqNum 1,
 protocolIdentifier { 0 0 8 2250 0 2 },
 discoveryComplete FALSE,
 callSignalAddress
 {
  ipAddress :
  {
   ip 'AC1301C8'H,
   port 1720
  }
 },
 rasAddress
 {
  ipAddress :
  {
   ip 'AC1301C8'H,
   port 22000
  }
 },
 terminalType
 {
  gateway
  {
},
  mc FALSE,
```

call Trace

- RAS brief/detail (2/2)–

```
undefinedNode FALSE
 },
terminalAlias
 {
  h323-ID : "jykim-2000",
  e164 : "2000"
 },
 endpointVendor
 {
  vendor
  {
   t35CountryCode 97,
   t35Extension 0,
   manufacturerCode 22
  },
  productId '41646450616320566F4950'H,
  versionId '382E3130'H
 },
 timeToLive 60,
 keepAlive FALSE,
 willSupplyUUIEs FALSE
}
1088942155 :  Sending RAS PDU to (172.19.1.248:1719) from 22000
 0e 80 00 00 06 00 08 91  4a 00 02 00 01 00 ac 13   ........J.......
 01 c8 06 b8 01 00 ac 13  01 c8 55 f0 08 00 02 40   ..........U....@
 09 00 6a 00 79 00 6b 00  69 00 6d 00 2d 00 32 00   ..j.y.k.i.m.-.2.
 30 00 30 00 30 01 80 53  33 60 61 00 00 16 0a 41   0.0.0..S3`a....A
 64 64 50 61 63 20 56 6f  49 50 03 38 2e 31 30 0e   ddPac VoIP.8.10.
 8a 02 00 3b 01 00 01 00
...:....
```

```
AP200(config-gateway)# <> RasMessage Decoding ... - Buffer(size:119)
value RasMessage ::= registrationConfirm :
{
 requestSeqNum 1,
 protocolIdentifier { 0 0 8 2250 0 2 },
 callSignalAddress
 {
  ipAddress :
  {
   ip 'AC1301C8'H,
   port 1720
  }
 },
 terminalAlias
 {
  h323-ID : "jykim-2000",
  e164 : "2000"
 },
 gatekeeperIdentifier "addpac-private-gk1",
 endpointIdentifier "40ae783c000081",
 timeToLive 60,
 willRespondToIRR FALSE
}
1088942155 :  Received RAS PDU from (172.19.1.248:1719)
 12 c0 00 00 06 00 08 91  4a 00 02 01 00 ac 13 01   ........J.......
 c8 06 b8 02 40 09 00 6a  00 79 00 6b 00 69 00 6d   ....@..j.y.k.i.m
 00 2d 00 32 00 30 00 30  00 30 01 80 53 33 22 00   .-.2.0.0.0..S3".
 61 00 64 00 64 00 70 00  61 00 63 00 2d 00 70 00   a.d.d.p.a.c.-.p.
 72 00 69 00 76 00 61 00  74 00 65 00 2d 00 67 00   r.i.v.a.t.e.-.g.
....:..
```

call Trace

- H323 Call Setup brief (1/2)–

```
AP200# debug voip call
AP200# 1      <CEP    000000> : Call Received
2     <CEP    000000> : Call Initiated : calledNumber() callerInfo() crv(0)
          total(0)
3     <Call   346>   : ****************** Call Created
          status(InitiatedByFXS) ******************
*
4     <CEP    000000> : Call id(50f3e740-9c8f-bbcc-81cb-0002a400380d)
          callNum(346)
5     <Call   346>   : MatchAllProcess After Sorted
                   <0>  id(4000) dest(....) prefer(0) selected(120)
                   <1>  id(1000) dest(T) prefer(0) selected(109)
6     <Call   346>   : Initiate callee with dial-peer(....)
          status(CalleeDeterminedAll) id(50f3e74
0-9c8f-bbcc-81cb-0002a400380d)
7     <NetEP  346>    : InitiateOutCall: calledNum(#2000) callingNum(2000)
          target(ras)
8     <NetEP  346>    : DoCall: calledAddr(#2000@) callingAddr(2000)
9     <GK     346>    : Send ARQ.
10    <GK     346>    : Received ACF.
11    <H225   346>    : Try signalling TCP connect (172.19.1.250:1720)
12    <H225   346>    : Signalling TCP connect success (346)
13    <H323   346>    : local capabilities.
                   number of capabilities = 5
                         1 : g7231A-6.3k
                         2 : g729-8k
                         3 : T.38
                         4 : UserInput/basicString
                         5 : UserInput/hookflash
14    <Q931   346>    : Send SETUP
15    <Q931   346>    : Received CALL PROCEEDING
```

```
16    <Chan   346>    : Open - number(101) direction(receive)
          session(voice) codec(g7231A-6.3k)
                   - Local : Data(23002) Cont(23003) Addr(172.19.1.200)
                   - Remote : Data(23900) Cont(23901)
          DataAddr(172.19.1.250) ContAddr(172.19.1
.250)
17    <Q931   346>    : Received PROGRESS
18    <Q931   346>    : Received Progress Indicator : desc(8), coding(0),
          location(0)
19    <H245   346>    : Send TCS request.
20    <Q931   346>    : Send FACILITY
21    <H245   346>    : Send MSD request.
22    <Q931   346>    : Send FACILITY
23    <Call   346>    : Progress from(ffffffff)
24    <Q931   346>    : Received CONNECT
25    <H225   346>    : Remote Endpoint (AddPac VoIP,8.00,97,0,22)
26    <Call   346>    : Connected from(ffffffff)
27    <NetEP  346>    : Call with voip.172.19.1.250 established
28    <Chan   346>    : Open - number(101) direction(transmit)
          session(voice) codec(g7231A-6.3k)
                   - Local : Data(23002) Cont(23003) Addr(172.19.1.200)
                   - Remote : Data(23900) Cont(23901)
          DataAddr(172.19.1.250) ContAddr(172.19.1
.250)
29    <Q931   346>    : Received FACILITY
30       <H245   346>    : Received TCS request.
```

call Trace

- H323 Call Setup brief (2/2) –

```
31   <H245   346>   : remote capabilities matching to local capabilities.
                      number of capabilities = 5
                         1 : g7231A-6.3k
                         2 : g729-8k
                         3 : T.38
                         4 : UserInput/basicString
                         5 : UserInput/hookflash
32   <H245   346>   : Send TCS ack.
33   <Q931   346>   : Send FACILITY
34   <Q931   346>   : Received FACILITY
35   <H245   346>   : Received MSD request.
36   <H245   346>   : Send MSD ack.
37   <Q931   346>   : Send FACILITY
38   <Q931   346>   : Received FACILITY
39   <H245   346>   : Received TCS ack..
40   <Q931   346>   : Received FACILITY
41   <H245   346>   : Received MSD ack.
42   <CEP    000000> : Disconnected(16)
43   <Call   346>   : Terminated  from(0) this(Local:CallClear)
       before(NULL) forced(0)
44   <Chan   346>   : Close - number(101) direction(receive)
45   <Chan   346>   : Close - number(101) direction(transmit)
46   <Q931   346>   : Send RELEASE COMPLETE
47   <GK     346>   : Send DRQ.
48   <GK     346>   : Received DCF.
49     <NetEP  346>   : Call TO <voip.172.19.1.250> terminated
       reason(Local:CallClear)
```

BLANK

call Trace

- h245 DTMF Relay brief/detail–

```
AP200# debug voip call
55      <H245   347>    : Send UserInputIndication(1)
56      <Q931   347>    : Send FACILITY
57      <Call   347>    : Digit(2) s(Established)
58      <H245   347>    : Send UserInputIndication(2)
59      <Q931   347>    : Send FACILITY
60      <Call   347>    : Digit(3) s(Established)
61      <H245   347>    : Send UserInputIndication(3)
62      <Q931   347>    : Send FACILITY
63      <Call   347>    : Digit(4) s(Established)
64      <H245   347>    : Send UserInputIndication(4)
65      <Q931   347>    : Send FACILITY
66      <Call   347>    : Digit(5) s(Established)
67      <H245   347>    : Send UserInputIndication(5)
68          <Q931   347>    : Send FACILITY
```

```
AP200# debug voip h245
AP200# <> MultimediaSystemControlMessage(choice:4)] Encoding ...
value MultimediaSystemControlMessage ::= indication : userInput :
alphanumeric : "1"
1088944039 :  Sent H245 PDU to (172.19.1.250:10091) from 18001

<> MultimediaSystemControlMessage(choice:4)] Encoding ...
value MultimediaSystemControlMessage ::= indication : userInput :
alphanumeric : "2"
1088944040 :  Sent H245 PDU to (172.19.1.250:10091) from 18001

<> MultimediaSystemControlMessage(choice:4)] Encoding ...
value MultimediaSystemControlMessage ::= indication : userInput :
alphanumeric : "3"
1088944041 :  Sent H245 PDU to (172.19.1.250:10091) from 18001

<> MultimediaSystemControlMessage(choice:4)] Encoding ...
value MultimediaSystemControlMessage ::= indication : userInput :
alphanumeric : "4"
1088944042 :  Sent H245 PDU to (172.19.1.250:10091) from 18001

<> MultimediaSystemControlMessage(choice:4)] Encoding ...
value MultimediaSystemControlMessage ::= indication : userInput :
alphanumeric : "5"
1088944042 :  Sent H245 PDU to (172.19.1.250:10091) from 18001

<> MultimediaSystemControlMessage(choice:3)] Encoding ...
value MultimediaSystemControlMessage ::= command :
endSessionCommand : disconnect : NULL
1088944045 :  Sent H245 PDU to (172.19.1.250:10091) from 18001
```

call Trace

- SIP Register(1/2)–

```
AP1100# debug voip sip
AP1100# debug voip call
 Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
REGISTER sip:test.sip.0038.net SIP/2.0
Date: Sun,27 Jul 2003 03:06:10 GMT
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 100.1.1.40:5060;branch=z9hG4bKa23f2000a41
Expires: 3600
From: sip:815000380100@test.sip.0038.net;tag=a23f2000a4
Call-ID: a241233f-7bac-2072-8000-0002a4ffff1a@100.1.1.40
To: sip:815000380100@test.sip.0038.net
Contact: sip:815000380100@100.1.1.40
Content-Length: 0
User-Agent: AddPac SIP Gateway ver 6.06
Max-Forwards: 70


        Received SIP PDU from ( 100.1.1.228:5060 )
SIP/2.0 401 Unauthorized
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 100.1.1.40:5060;branch=z9hG4bKa23f2000a41
From: sip:815000380100@test.sip.0038.net;tag=a23f2000a4
Call-ID: a241233f-7bac-2072-8000-0002a4ffff1a@100.1.1.40
To: sip:815000380100@test.sip.0038.net
Content-Length: 0
Server: LMAP-LongBoard Inc.
WWW-Authenticate: Digest realm="Registered
             Users",qop="auth",opaque="12510306183816
29",nonce="1251030618381629"
```

```
 9      <SIP    1>    : Receive 401  Unauthorized

10      <SIP    1>    : Transaction (1 REGISTER) completed
11      <SIP    0>    : Adding authentication information
12      <SIP    2>    : Send REGISTER Request

        Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
REGISTER sip:test.sip.0038.net SIP/2.0
Date: Sun,27 Jul 2003 03:06:12 GMT
CSeq: 2 REGISTER
Via: SIP/2.0/UDP 100.1.1.40:5060;branch=z9hG4bKa23f2000a42
Expires: 3600
Authorization: Digest username="815000380100", realm="Registered Users",
             nonce="125
1030618381629", opaque="1251030618381629", uri="sip:test.sip.0038.net",
             response="f
36127731b8062c0dfb0f146c97651ee", algorithm=MD5
From: sip:815000380100@test.sip.0038.net;tag=a23f2000a4
Call-ID: a241233f-7bac-2072-8000-0002a4ffff1a@100.1.1.40
To: sip:815000380100@test.sip.0038.net
Contact: sip:815000380100@100.1.1.40
Content-Length: 0
User-Agent: AddPac SIP Gateway ver 6.06
Max-Forwards: 70
```

call Trace

- SIP Register (2/2) –

Received SIP PDU from (100.1.1.228:5060)
SIP/2.0 200 Ok
CSeq: 2 REGISTER
Via: SIP/2.0/UDP 100.1.1.40:5060;branch=z9hG4bKa23f2000a42
From: sip:815000380100@test.sip.0038.net;tag=a23f2000a4
Call-ID: a241233f-7bac-2072-8000-0002a4ffff1a@100.1.1.40
To: sip:815000380100@test.sip.0038.net
Content-Length: 0
Server: LMAP-LongBoard Inc.
Contact: <sip:815000380100@100.1.1.40>;q=0.500;expires=3600

13 <SIP 2> : Receive 200 Ok

14 <SIP 2> : Transaction (2 REGISTER) completed

Blank

call Trace

- SIP Setup (1/2)–

```
AP1100# debug voip sip
AP1100# debug voip call
AP1100# 1       <Time   7>      : Inter digit timer timeout.
2       <Call   7>      : Digit(#)
3       <Call   7>      : MatchAllProcess After Sorted
                          <0>  id(1000) dest(T) prefer(0) selected(1)
4       <Call   7>      : Initiate callee with dial-peer(T)
status(CalleeDetermined
All)
5       <NetEP  7>      : InitiateOutCall: calledNum(011177)
callingNum(81500038010
0) target(sip-server)
6       <NetEP  7>      : DoCall:
calledAddr(sip:011177@test.sip.0038.net) callingA
ddr(815000380100)
7       <SIP    0>      : No authentication information available
8       <SIP    7>      : Send INVITE Request

        Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
INVITE sip:011177@test.sip.0038.net SIP/2.0
Date: Sun,27 Jul 2003 02:18:55 GMT
CSeq: 3 INVITE
Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a43
From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
To: <sip:011177@test.sip.0038.net>
Contact: sip:815000380100@218.47.24.3
Content-Type: application/sdp
Content-Length: 145
User-Agent: AddPac SIP Gateway ver 6.06
Max-Forwards: 70
```

```
v=0
o=AddpacTechSIP-GW-UserAgent 1059272335 1059272335 IN
IP4 218.47.24.3
s=SIP Session
t=0 0
c=IN IP4 218.47.24.3
m=audio 23010 RTP/AVP 0

        Received SIP PDU from ( 100.1.1.228:5060 )
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a43
CSeq: 3 INVITE
From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
To: <sip:011177@test.sip.0038.net>

9       <SIP    7>      : Receive 100  Trying

10      <SIP    7>      : Transaction (3 INVITE) proceeding

```

call Trace

- SIP Setup (2/2)–

```
68      Received SIP PDU from ( 100.1.1.228:5060 )
69      SIP/2.0 100 Trying
70      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a44
71      CSeq: 4 INVITE
72      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
73      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
74      To: <sip:011177@test.sip.0038.net>

75      15    <SIP    7>    : Receive 100  Trying

76      16    <SIP    7>    : Transaction (4 INVITE) proceeding

77          Received SIP PDU from ( 100.1.1.228:5060 )
78      SIP/2.0 180 Ringing
79      CSeq: 4 INVITE
80      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a44
81      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
82      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
83      To: <sip:011177@test.sip.0038.net>;tag=apsd57595u1-7881812
84      Content-Length: 159
85      Content-Disposition: signal;handling=required
86      Content-Type: application/sdp

87      v=0
88      o=Sonus_UAC 32590 29093 IN IP4 100.1.1.228
89      s=SIP Media Capabilities
90      c=IN IP4 100.1.1.228
91      t=0 0
92      m=audio 10008 RTP/AVP 0
93      a=sendrecv
94      a=ptime:10
```

```
68      a=sendrecv
69      a=ptime:10
       Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
68      17    <SDP    0>    : Malformed media attribute : sendrecv
69      18    <SDP    0>    : Malformed media attribute : ptime:10
70      19    <SIP    7>    : Receive 180  Ringing

71      20    <SIP    7>    : Transaction (4 INVITE) proceeding
72      21    <Call   7>    : Alert from(fffffffe) pseudo(0) inband(0)
        status(CalleeIni
73      tiated)

74          Received SIP PDU from ( 100.1.1.228:5060 )
75      SIP/2.0 200 OK
76      CSeq: 4 INVITE
77      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a44
78      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
79      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
80      To: <sip:011177@test.sip.0038.net>;tag=apsd57595u1-7881812
81      Contact: <sip:011177-bco-
        r1eed7gso8jme@100.1.1.228:5060;transport=udp>
82      Allow: OPTIONS,INVITE, CANCEL, ACK, PRACK, INFO, BYE
83      Accept: multipart/mixed, application/sdp, application/isup,
        application/dtmf, appli
84      cation/dtmf-relay
85      Content-Length: 159
86      Content-Disposition: signal;handling=required
87      Content-Type: application/sdp
```

call Trace

- SIP Setup (2/2)–

```
68      v=0
69      o=Sonus_UAC 32590 29093 IN IP4 100.1.1.228
70      s=SIP Media Capabilities
71      c=IN IP4 100.1.1.228
72      t=0 0
73      m=audio 10008 RTP/AVP 0
74      a=sendrecv
75      a=ptime:10

76      22    <SDP    0>    : Malformed media attribute : sendrecv
77      23    <SDP    0>    : Malformed media attribute : ptime:10
78      24    <SIP    7>    : Receive 200  OK

79      25    <SIP    7>    : Send ACK Request

80
```

```
68      Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
69      ACK sip:011177@test.sip.0038.net SIP/2.0
70      CSeq: 4 ACK
71      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a44
72      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
73      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
74      To: <sip:011177@test.sip.0038.net>;tag=apsd57595u1-7881812
75      Content-Length: 0
76      Max-Forwards: 70

77      26    <SIP    7>    : Get SIP MediaFormat : 0
78      27    <Call   7>    : Connected from(fffffffe)
79      28    <NetEP  7>    : Call with sip:011177@test.sip.0038.net
        established
80      29    <SIP    7>    : Received INVITE OK response
81      30    <Time   7>    : SIP_TCOMP_LONG timer timeout.
        (transactionID : 185270320)
82      31    <Time   7>    : SIP_TCOMP_LONG timer timeout.
        (transactionID : 185270336
```

call Trace

- SIP Setup (2/2)–

```
68      Received SIP PDU from ( 100.1.1.228:5060 )
69      SIP/2.0 407 Proxy Authentication Required
70      CSeq: 3 INVITE
71      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a43
72      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
73      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
74      To: <sip:011177@test.sip.0038.net>;tag=aprqngfrt-apsd57595u1-
        z9hG4bK219842720
75      Content-Length: 0
76      Server: LMAP-LongBoard Inc.
77      Proxy-Authenticate: Digest realm="Registered
        Users",qop="auth",opaque="109103060351
78      1529",nonce="1091030603511529"

79      11    <SIP    7>    : Receive 407  Proxy Authentication Required

80      12    <SIP    7>    : Send ACK Request

81          Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
82      ACK sip:011177@test.sip.0038.net SIP/2.0
83      CSeq: 3 ACK
84      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a43
85      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
86      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
87      To: <sip:011177@test.sip.0038.net>;tag=aprqngfrt-apsd57595u1-
        z9hG4bK219842720
88      Content-Length: 0
89      Max-Forwards: 70

90      13    <SIP    0>    : Adding authentication information
91      14    <SIP    7>    : Send INVITE Request
```

```
        Sending SIP PDU to ( test.sip.0038.net:5060 ) from 5060
68      INVITE sip:011177@test.sip.0038.net SIP/2.0
69      Date: Sun,27 Jul 2003 02:18:55 GMT
70      CSeq: 4 INVITE
71      Via: SIP/2.0/UDP 218.47.24.3:5060;branch=z9hG4bK8f3f5910a44
72      From: <sip:815000380100@test.sip.0038.net>;tag=8f3f5910a4
73      Call-ID: 8f36233f-ecdb-5987-8010-0002a4ffff1a@218.47.24.3
74      To: <sip:011177@test.sip.0038.net>
75      Contact: sip:815000380100@218.47.24.3
76      Proxy-Authorization: Digest username="815000380100",
        realm="Registered Users", nonc
77      e="1091030603511529", opaque="1091030603511529",
        uri="sip:011177@test.sip.0038.net"
78      , response="2077b54acf82b8ed35967199bf38023e",
        algorithm=MD5
79      Content-Type: application/sdp
80      Content-Length: 145
81      User-Agent: AddPac SIP Gateway ver 6.06
82      Max-Forwards: 70

83      v=0
84      o=AddpacTechSIP-GW-UserAgent 1059272335 1059272335 IN
        IP4 218.47.24.3
85      s=SIP Session
86      t=0 0
87      c=IN IP4 218.47.24.3
88      m=audio 23010 RTP/AVP 0
```

call Trace

- T38 FAX Relay (1/4) –

```
AP200# debug voip call
AP200# debug rta ipc
AP200# debug rta t38
AP200# [1268.205] VM(0/0/0) vmOffHook
[1268.265] VM(0/0/0) vmTmoOffHook
[1268.265] VM(0/0/0) Rx OffHook
[1268.265] VM(0/0/0) Tx OFFHOOK_IND
[1268.265] VM(0/0/0) play Dial tone
1       <CEP    000000> : Call Received
2       <CEP    000000> : Call Initiated : calledNumber() callerInfo() crv(0)
            total(0)
3       <Call   4>      : ****************** Call Created status(InitiatedByFXS)
            ****************
***
4       <CEP    000000> : Call id(b06eec40-e399-77ad-8007-
            0002a400e070) callNum(4)
[1271.575] VM(0/0/0) Tx DIGIT_IND '1'
[1271.575] VM(0/0/0) play mute
5       <Call   4>      : Digit(1) s(InitiatedByFXS)
6       <Call   4>      : Digit match checked(MatchedPartially)
[1271.755] VM(0/0/0) Tx DIGIT_IND '0'
7       <Call   4>      : Digit(0) s(CalleeUndetermined)
8       <Call   4>      : Digit match checked(MatchedPartially)
[1271.935] VM(0/0/0) Tx DIGIT_IND '0'
9       <Call   4>      : Digit(0) s(CalleeUndetermined)
10      <Call   4>      : Digit match checked(MatchedPartially)
[1272.115] VM(0/0/0) Tx DIGIT_IND '0'
11      <Call   4>      : Digit(0) s(CalleeUndetermined)
12      <Call   4>      : Digit match checked(MatchedPerfect)
13      <Call   4>      : MatchAllProcess After Sorted
                        <0>  id(1000) dest(1000) prefer(0) selected(2)
```

```
14      <Call   4>      : Initiate callee with dial-peer(1000)
            status(CalleeDeterminedAll) id(b06ee
c40-e399-77ad-8007-0002a400e070)
15      <NetEP  4>      : InitiateOutCall: calledNum(1000) callingNum(2000)
            target(172.17.180.3)
16      <NetEP  4>      : DoCall: calledAddr(1000@172.17.180.3)
            callingAddr(2000)
[1272.120] VM(0/0/0) Fax rate  9600
17      <H225   4>      : Try signalling TCP connect (172.17.180.3:1720)
18      <H225   4>      : Signalling TCP connect success (4)
19      <H323   4>      : local capabilities.
                    number of capabilities = 5
                        1 : g7231A-6.3k
                        2 : g729-8k
                        3 : T.38
                        4 : UserInput/basicString
                        5 : UserInput/hookflash
20      <Q931   4>      : Send SETUP
[1272.140] RTA(0/0/0) Rx RS_LISTEN_REQ callId=4 ssId=1 G729A
     peer=0.0.0.0 mp=23006/23007 hp=0/0
[1272.140] VM(0/0/0) codec replace later
21      <Q931   4>      : Received CALL PROCEEDING
22      <Chan   4>      : Open - number(101) direction(receive)
            session(voice) codec(g7231A-6.3k)
                    - Local : Data(23006) Cont(23007) Addr(172.17.180.2)
                    - Remote : Data(23006) Cont(23007)
            DataAddr(172.17.180.3) ContAddr(172.17
.180.3)
23      <Q931   4>      : Received ALERTING
24      <H245   4>      : Send TCS request.
25      <Q931   4>      : Send FACILITY
```

call Trace

- T38 FAX Relay (2/4) –

```
26      <H245   4>      : Send MSD request.
27      <Q931   4>      : Send FACILITY
28      <Call   4>     : Alert from(ffffffff) pseudo(0) inband(0)
            status(CalleeInitiated)
[1272.225] RTA(0/0/0) Rx CC_ALERT_RSP peerId(0/0/0)
[1272.225] VM(0/0/0) play RingBack tone
29      <Q931   4>      : Received FACILITY
30      <H245   4>      : Received TCS request.
31      <H245   4>      : remote capabilities matching to local capabilities.
                      number of capabilities = 5
                             1 : g7231A-6.3k
                             2 : g729-8k
                             3 : T.38
                             4 : UserInput/basicString
                             5 : UserInput/hookflash
32      <H245   4>      : Send TCS ack.
33      <Q931   4>      : Send FACILITY
34      <Q931   4>      : Received FACILITY
35      <H245   4>      : Received TCS ack..
36      <Q931   4>      : Received FACILITY
37      <H245   4>      : Received MSD ack.
38      <H245   4>      : Send MSD ack.
39      <Q931   4>      : Send FACILITY
40      <Q931   4>      : Received CONNECT
41      <H225   4>      : Remote Endpoint (AddPac VoIP,7.00,97,0,22)
42      <Call   4>     : Connected from(ffffffff)
[1279.375] RTA(0/0/0) Rx CC_CONNECT_RSP peerId(0/0/0)
[1279.375] VM(0/0/0) Fax enable
[1279.375] VM(0/0/0) play mute
43      <NetEP  4>      : Call with voip.172.17.180.3 established
```

```
44      <Chan   4>      : Open - number(101) direction(transmit)
            session(voice) codec(g7231A-6.3k)
                     - Local : Data(23006) Cont(23007) Addr(172.17.180.2)
                     - Remote : Data(23006) Cont(23007)
            DataAddr(172.17.180.3) ContAddr(172.17
.180.3)
[1279.375] RTA(0/0/0) Rx RS_OPEN_REQ callId=4 ssId=1 G7236
      peer=172.17.180.3 mp=23006/23007 hp=23006/23007
[1279.380] VM(0/0/0) vopp idle
[1279.380] VM(0/0/0) codec replace timer start
[1279.385] VM(0/0/0) discard voice under codec replace
[1279.395] VM(0/0/0) discard voice under codec replace
[1279.400] VM(0/0/0) under codec replace
[1279.440] VM(0/0/0) codec replaced to G7236
[1279.440] VM(0/0/0) Fax enable
[1279.440] VM(0/0/0) play mute
45      <Q931   4>      : Received FACILITY
46      <H245   4>      : Received RM request.
47      <Q931   4>      : Send FACILITY
48      <H245   4>      : Send CLC request.
[1284.370] RTA(0/0/0) Rx RS_CLOSE_REQ callId=4 ssId=1 dir=forw
49      <Chan   4>      : Close - number(101) direction(transmit)
50      <Q931   4>      : Send FACILITY
51      <H245   4>      : Send OLC request.
52      <Q931   4>      : Send FACILITY
53      <Q931   4>      : Received FACILITY
54      <H245   4>      : Received CLC request.
55      <H245   4>      : Send CLC ack.
56      <Q931   4>      : Send FACILITY
[1284.395] RTA(0/0/0) Rx RS_CLOSE_REQ callId=4 ssId=1 dir=reve
```

call Trace

- T38 FAX Relay (3/4) –

```
57    <Chan   4>    : Close – number(101) direction(receive)
58    <Q931   4>    : Received FACILITY
59    <H245   4>    : Received OLC request.
60    <Chan   4>    : Open – number(102) direction(receive) session(data)
          codec(T.38)
                    – Local : Data(23006) Cont(23007) Addr(172.17.180.2)
                    – Remote : Data(23006) Cont(23007)
          DataAddr(172.17.180.3) ContAddr(172.17
.180.3)
[1284.405] RTA(0/0/0) Rx RS_LISTEN_REQ callId=4 ssId=3 G711A
        peer=172.17.180.3 mp=23006/23007 hp=23006/23007
61    <H245   4>    : Send OLC ack.
62    <Q931   4>    : Send FACILITY
63    <Q931   4>    : Received FACILITY
64    <H245   4>    : Received CLC ack.
65    <Q931   4>    : Received FACILITY
66    <H245   4>    : Received OLC ack.
67    <Chan   4>    : Open – number(103) direction(transmit) session(data)
          codec(T.38)
                    – Local : Data(23006) Cont(23007) Addr(172.17.180.2)
                    – Remote : Data(23006) Cont(23007)
          DataAddr(172.17.180.3) ContAddr(172.17
.180.3)
[1284.450] RTA(0/0/0) Rx RS_OPEN_REQ callId=4 ssId=3 G711A
        peer=172.17.180.3 mp=23006/23007 hp=23006/23007
[1284.495] T38(0/0/0) Rx T30I CED
[1291.185] T38(0/0/0) Rx T30I FLAG
[ff] [c0] [02] [04] [04] [04] [04] [04] [04] [04] [04] [04] [04] [04]
          [04] [04] [04] [04] [04] [04]
 [1e] [86] [62] [1293.520] T38(0/0/0) Rx T30D V21 OK
[ff] [c8] [01] [00] [73] [1e] [1293.730] T38(0/0/0) Rx T30D V21 OK
[1293.760] T38(0/0/0) Rx T30D V21 END
```

```
[1294.390] T38(0/0/0) Tx T30I FLAG
[1296.010] T38(0/0/0) Tx HDLC L=23: ff c0 c2 04 04 04 04 04 04 04 04
          04 04 04 04 04 04 04 04
04 04
[1296.015] T38(0/0/0) Tx T30D V21 OK
[1296.285] T38(0/0/0) Tx HDLC L=6: ff c8 c1 00 61 18
[1296.290] T38(0/0/0) Tx T30D V21 OK
[1296.370] T38(0/0/0) Tx T30D V21 END
[1296.490] T38(0/0/0) Tx T30I V29_9600
iiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiii[1298.190] T38(0/0/0) Tx T30D V29_9600 T4_E
[1299.370] T38(0/0/0) Rx T30I FLAG
[ff] [c8] [21] [1300.600] T38(0/0/0) Rx T30D V21 OK
[1300.630] T38(0/0/0) Rx T30D V21 END
[1301.530] T38(0/0/0) Tx T30I V29_9600
iiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiii
iiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiiii[1309.115] T
38(0/0/0) Tx T30D V29_9600 T4_E
[1309.270] T38(0/0/0) Tx T30I FLAG
[1310.375] T38(0/0/0) Tx HDLC L=3: ff c8 f4
[1310.375] T38(0/0/0) Tx T30D V21 OK
[1310.440] T38(0/0/0) Tx T30D V21 END
[1311.130] T38(0/0/0) Rx T30I FLAG
[ff] [c8] [31] [1312.360] T38(0/0/0) Rx T30D V21 OK
[1312.390] T38(0/0/0) Rx T30D V21 END
[1312.990] T38(0/0/0) Tx T30I FLAG
[1314.085] T38(0/0/0) Tx HDLC L=3: ff c8 df
[1314.085] T38(0/0/0) Tx T30D V21 OK
[1314.160] T38(0/0/0) Tx T30D V21 END
68    <Q931   4>    : Received RELEASE COMPLETE
[1315.030] RTA(0/0/0) Rx RS_CLOSE_REQ callId=4 ssId=3 dir=reve
69    <Chan   4>    : Close – number(102) direction(receive)
```

call Trace

- T38 FAX Relay (4/4) –

```
[1315.030] RTA(0/0/0) Rx RS_CLOSE_REQ callId=4 ssId=3 dir=forw
70      <Chan   4>       : Close - number(103) direction(transmit)
71      <Call   4>       : Terminated  from(ffffffff) this(Remote:CallClear)
            before(NULL) forced(0)
72      <CEP    000000> : Disconnect (0)
[1315.085] RTA(0/0/0) Rx CC_DISCONN_REQ CZ=0, peerId(0/0/0)
[1315.085] VM(0/0/0) play Reorder tone
73      <NetEP  4>       : Call TO <voip.172.17.180.3> terminated
            reason(Remote:CallClear)
[1315.185] VM(0/0/0) vmOnHook
[1315.245] VM(0/0/0) vmTmoOnHook
[1315.305] VM(0/0/0) vmTmoOnHook
[1315.365] VM(0/0/0) vmTmoOnHook
[1315.425] VM(0/0/0) vmTmoOnHook
[1315.485] VM(0/0/0) vmTmoOnHook
[1315.545] VM(0/0/0) vmTmoOnHook
[1315.605] VM(0/0/0) vmTmoOnHook
[1315.665] VM(0/0/0) vmTmoOnHook
[1315.725] VM(0/0/0) vmTmoOnHook
[1315.785] VM(0/0/0) vmTmoOnHook
[1315.845] VM(0/0/0) vmTmoOnHook
[1315.845] VM(0/0/0) Rx OnHook
[1315.845] VM(0/0/0) vopp idle
[1315.845] VM(0/0/0) Tx DISCONN_CNF
74      <CEP    000000> : Disconnected(16)
```

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call Trace

- E&M –

```
AP2520# debug voip call
AP2520# debug rta ipc
AP2520# [76.030] VM(1/0/0) vmOffHook
[76.030] VM(1/0/0) Rx OffHook
[76.030] VM(1/0/0) Tx CALL_RECEIVED
1       <CEP    010000> : Call Received
[76.230] VM(1/0/0) WinkWait timeout
[76.230] VM(1/0/0) E&M OffHook
[76.430] VM(1/0/0) WinkDura timeout, Tx OFFHOOK_IND
[76.430] VM(1/0/0) E&M OnHook
[76.430] VM(1/0/0) play Dial tone
2       <CEP    010000> : Call Initiated : calledNumber() crv(0) total(0)
3           <Call   1>      : ****************** Call Created
            status(InitiatedByE&M)  *******
```

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call Trace

- ISDN PRI (1/3)–

```
AP2520# debug voip call
AP2520# debug rta ipc
AP2520# debug rta q931
AP2520# [6425.650] Q931[0] Rx DL_DATA_IND len=37
[6425.650] Q931[0] Rx [SETUP] 08 02 0e a3 05 a1 04 03 80 90 a3 18 03
        a9 83 8b 6c 05 c9 32 3
0 30 33 70 0c 80 35 36 36 34 37 37 35 31 30 30 30
 SendingCom : a1
 Bearer_Cap : 04 03 80 90 a3
  ChannelId : 18 03 a9 83 8b
 CallingNum : 6c 05 c9 32 30 30 33
  CalledNum : 70 0c 80 35 36 36 34 37 37 35 31 30 30 30
1     <Dev    000000> : CCC_GetBchannel :  bChNo(11) exclusive(1)
[6425.655] Q931[0] Tx PCC_SETUP_IND CR=3747 B=11 Excl CZ=0 PG=0
        Cd='56647751000' Cg='2003'
[6425.655] Q931[0] Tx [CALL_P] 08 02 8e a3 02 18 03 a9 83 8b
  ChannelId : 18 03 a9 83 8b
2     <CEP    00000b> : Call Received
3     <Call   34>     : ****************** Call Created status(InitiatedByE1)
        *********
**********
4     <CEP    00000b> : Calling number(2003)
5     <CEP    00000b> : Call id(3b06ec40-73f5-7768-8033-0002a4005670)
        callNum(34)
6     <Call   34>     : Match check (MatchedPerfect)
7     <Call   34>     : MatchAllProcess After Sorted
                <0>  id(0) dest(5664775....) prefer(0) selected(15)
8     <CGrp   000000> : GetAvailableChannel (15)
9     <Call   34>     : Initiate callee with dial-peer(5664775....)
        status(CalleeDetermin
edAll) id(3b06ec40-73f5-7768-8033-0002a4005670)
```

```
10     <CEP    00000f> : InitiateOutCall :  calledNum(1000),
callingNum(2003), callerPort(
b) type(E1)
[6425.665] RTA(0/0/11) Rx PCC_BCH_CONN peerId(0/0/15)
[6425.665] VM(0/0/11) Fax disable
[6425.665] VM(0/0/15) Fax disable
[6425.665] VM(0/0/11) play mute
[6425.665] Q931[0] Rx PCC_SETUP_REQ CR=16 B=15 Excl CZ=128 PG=0
Cd='1000' Cg=80 80 '2003'
[6425.665] Q931[0] Tx [SETUP] 08 02 00 10 05 04 03 80 90 a3 18 03 a9
83 8f 6c 05 80 32 30 3
0 33 70 05 80 31 30 30 30 7d 02 91 81 a1
 Bearer_Cap : 04 03 80 90 a3
  ChannelId : 18 03 a9 83 8f
 CallingNum : 6c 05 80 32 30 30 33
  CalledNum : 70 05 80 31 30 30 30
 HighLayerC : 7d 02 91 81
 SendingCom : a1
11     <CEP    00000f> : Outbound call to CEP callId(3b06ec40-73f5-
7768-8033-0002a4005670)
 callNum(34)
[6425.840] Q931[0] Rx DL_DATA_IND len=10
[6425.840] Q931[0] Rx [CALL_P] 08 02 80 10 02 18 03 a9 83 8f
  ChannelId : 18 03 a9 83 8f
[6425.840] Q931[0] Tx PCC_SETUP_CNF CR=16 B=15 Excl CZ=0 PG=0
Cd='1000' Cg=80 80 '2003'
[6425.840] RTA(0/0/15) Rx PCC_BCH_CONN peerId(0/0/11)
[6425.840] VM(0/0/15) Fax disable
[6425.840] VM(0/0/11) Fax disable
[6425.845] VM(0/0/15) play mute
[6425.855] Q931[0] Rx DL_DATA_IND len=10
```

call Trace

- ISDN PRI (2/3)–

```
[6425.855] Q931[0] Rx [ALERT] 08 02 80 10 01 18 03 a9 83 8f
  ChannelId : 18 03 a9 83 8f
[6425.855] Q931[0] Tx PCC_ALERT_IND CR=16 B=15 Excl CZ=0 PG=0
         Cd='1000' Cg=80 80 '2003'
12      <Call   34>     : Alert from(f) pseudo(0) inband(0)
         status(CalleeInitiated)
[6425.855] RTA(0/0/11) Rx CC_ALERT_RSP peerId(0/0/0)
[6425.855] VM(0/0/11) play RingBack tone
[6425.855] Q931[0] Rx PCC_ALERT_RSP CR=3747 B=11 Excl CZ=128
         PG=0
[6425.860] Q931[0] Tx [ALERT] 08 02 8e a3 01
[6429.685] Q931[0] Rx DL_DATA_IND len=5
[6429.685] Q931[0] Rx [CONNECT] 08 02 80 10 07
[6429.685] Q931[0] Tx PCC_CONN_CNF CR=16 B=15 Excl CZ=0 PG=0
         Cd='1000' Cg=80 80 '2003'
[6429.685] Q931[0] Tx [CONN_ACK] 08 02 00 10 0f
13      <Call   34>     : Connected from(f)
[6429.685] Q931[0] Rx PCC_CONN_REQ CR=3747 B=11 Excl CZ=128 PG=0
[6429.690] Q931[0] Tx [CONNECT] 08 02 8e a3 07
[6429.690] Q931[0] Tx PCC_CONN_CNF CR=3747 B=11 Excl CZ=0 PG=0
         Cd='56647751000' Cg='2003'
14      <Call   34>     : Connected from(b)
[6429.765] Q931[0] Rx DL_DATA_IND len=5
[6429.765] Q931[0] Rx [CONN_ACK] 08 02 0e a3 0f
[6431.405] Q931[0] Rx DL_DATA_IND len=13
[6431.405] Q931[0] Rx [DISCONN] 08 02 0e a3 45 08 02 81 90 1e 02 81
         88
    Cause : 08 02 81 90
 ProgressIn : 1e 02 81 88
[6431.405] Q931[0] Tx [RELEASE] 08 02 8e a3 4d 08 02 80 90
    Cause : 08 02 80 90
```

```
[6431.410] Q931[0] Tx PCC_DISC_IND CR=3747 B=11 Excl CZ=16 PG=8
Cd='56647751000' Cg='2003'
[6431.485] Q931[0] Rx DL_DATA_IND len=5
[6431.485] Q931[0] Rx [REL_COM] 08 02 0e a3 5a
[6431.485] Q931[0] Tx PCC_DISC_CNF CR=3747 B=11 Excl CZ=16 PG=8
Cd='56647751000' Cg='2003'
15      <CEP    00000b> : Disconnected(16)
[6431.485] RTA(0/0/11) Rx PCC_BCH_DISC peerId(0/0/0)
[6431.485] VM(0/0/11) vopp idle
16      <Call   34>     : Terminated  from(b) this(Local:CallClear)
before(NULL) forced(0)
17      <CEP    00000f> : StopSignal
[6431.490] RTA(0/0/15) Rx AP_SVC_REQ nSvcElem=1 rawDataLen=0
  DTMF_STOP
[6431.490] VM(0/0/15) play mute
18      <CEP    00000f> : Disconnect (0)
[6431.490] RTA(0/0/15) Rx PCC_BCH_DISC peerId(0/0/0)
[6431.490] VM(0/0/15) vopp idle
[6431.490] Q931[0] Rx PCC_DISC_REQ CR=16 B=15 Excl CZ=144 PG=0
[6431.490] Q931[0] Tx [DISCONN] 08 02 00 10 45 08 02 80 90
    Cause : 08 02 80 90
[6431.560] Q931[0] Rx DL_DATA_IND len=13
[6431.560] Q931[0] Rx [DISCONN] 08 02 80 10 45 08 02 81 90 1e 02 81
88
    Cause : 08 02 81 90
 ProgressIn : 1e 02 81 88
[6431.560] Q931[0] Tx [RELEASE] 08 02 00 10 4d 08 02 80 90
    Cause : 08 02 80 90
[6431.560] Q931[0] Tx PCC_DISC_IND CR=16 B=15 Excl CZ=16 PG=8
Cd='1000' Cg=80 80 '2003'
[6431.585] Q931[0] Rx DL_DATA_IND len=17
```

call Trace

- ISDN PRI (3/3)–

```
[6431.585] Q931[0] Rx [RELEASE] 08 02 80 10 4d 00 01 08 08 02 00 10 45
            08 02 80 90
 InfoElemId : 00 01 08
      Cause : 08 02 00 10
 InfoElemId : 45 08 02 80 90 50 3d 08 00 00
[6431.585] Q931[0] Tx PCC_DISC_CNF CR=16 B=15 Excl CZ=100 PG=8
            Cd='1000' Cg=80 80 '2003'
[6431.590] Q931[0] Tx [REL_COM] 08 02 00 10 5a 08 03 80 e4 08
      Cause : 08 03 80 e4 08
19          <CEP    00000f> : Disconnected(100)
```

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call Trace

- R2 (1/2) –

```
AP2520# debug v c
AP2520# debug rta r2
AP2520# debug rta ipc
AP2520# [4032.500] R2(0/0/12) Rx CAS A=0 B=0
[4032.500] R2(0/0/12) Tx CAS A=1 B=1
[4032.500] VM(0/0/12) Tx OFFHOOK_IND
1	<CEP	00000c> : Call Received
2	<CEP	00000c> : Call Initiated : calledNumber() crv(0) total(0)
3	<Call	7>	: ****************** Call Created status(InitiatedByE1)
	********
***********
4	<CEP	00000c> : Calling number()
[4032.505] VM(0/0/12) play mute
5	<CEP	00000c> : Call id(5405ec40-4b08-e229-800d-
	0002a400aca6) callNum(7)
[4035.700] R2(0/0/12) Rx FW I-5: Digit 5
[4035.700] VM(0/0/12) Tx DIGIT_IND '5'
[4035.700] R2(0/0/12) Tx BW A1: Send Next Digit
6	<Call	7>	: Digit(5) at InitiatedByE1
7	<Call	7>	: MatchedPartially
[4035.880] R2(0/0/12) MFC signal OFF, mute ON
[4035.880] VM(0/0/12) play mute
[4035.980] R2(0/0/12) mute timeout
[4036.000] R2(0/0/12) Rx FW I-6: Digit 6
[4036.000] VM(0/0/12) Tx DIGIT_IND '6'
[4036.000] R2(0/0/12) Tx BW A1: Send Next Digit
8	<Call	7>	: Digit(6) at CalleeUndetermined
9	<Call	7>	: MatchedPartially
[4036.180] R2(0/0/12) MFC signal OFF, mute ON
[4036.180] VM(0/0/12) play mute
[4036.280] R2(0/0/12) Rx FW I-6: Digit 6
```

```
[4036.280] VM(0/0/12) Tx DIGIT_IND '6'
[4036.280] R2(0/0/12) Tx BW A1: Send Next Digit
[4036.280] R2(0/0/12) mute timeout
10	<Call	7>	: Digit(6) at CalleeUndetermined
11	<Call	7>	: MatchedPartially
[4036.460] R2(0/0/12) MFC signal OFF, mute ON
[4036.460] VM(0/0/12) play mute
[4036.560] R2(0/0/12) mute timeout
[4036.580] R2(0/0/12) Rx FW I-4: Digit 4
[4036.580] VM(0/0/12) Tx DIGIT_IND '4'
[4036.580] R2(0/0/12) Tx BW A1: Send Next Digit
12	<Call	7>	: Digit(4) at CalleeUndetermined
13	<Call	7>	: MatchedPartially
[4036.760] R2(0/0/12) MFC signal OFF, mute ON
[4036.760] VM(0/0/12) play mute
[4036.860] R2(0/0/12) mute timeout
[4036.880] R2(0/0/12) Rx FW I-7: Digit 7
[4036.880] VM(0/0/12) Tx DIGIT_IND '7'
[4036.880] R2(0/0/12) Tx BW A1: Send Next Digit
14	<Call	7>	: Digit(7) at CalleeUndetermined
15	<Call	7>	: MatchedPartially
[4037.060] R2(0/0/12) MFC signal OFF, mute ON
[4037.060] VM(0/0/12) play mute
[4037.160] R2(0/0/12) mute timeout
[4037.170] R2(0/0/12) Rx FW I-7: Digit 7
[4037.170] VM(0/0/12) Tx DIGIT_IND '7'
[4037.170] R2(0/0/12) Tx BW A1: Send Next Digit
16	<Call	7>	: Digit(7) at CalleeUndetermined
17	<Call	7>	: MatchedPartially
[4037.350] R2(0/0/12) MFC signal OFF, mute ON
[4037.350] VM(0/0/12) play mute
```

call Trace

- R2 (2/2) –

```
[4037.450] R2(0/0/12) mute timeout
[4037.470] R2(0/0/12) Rx FW I-5: Digit 5
[4037.470] VM(0/0/12) Tx DIGIT_IND '5'
[4037.470] R2(0/0/12) Tx BW A1: Send Next Digit
18      <Call   7>      : Digit(5) at CalleeUndetermined
19      <Call   7>      : MatchedPartially
[4037.650] R2(0/0/12) MFC signal OFF, mute ON
[4037.650] VM(0/0/12) play mute
[4037.750] R2(0/0/12) mute timeout
[4037.770] R2(0/0/12) Rx FW I-1: Digit 1
[4037.770] VM(0/0/12) Tx DIGIT_IND '1'
[4037.770] R2(0/0/12) Tx BW A1: Send Next Digit
20      <Call   7>      : Digit(1) at CalleeUndetermined
21      <Call   7>      : MatchedPartially
[4037.950] R2(0/0/12) MFC signal OFF, mute ON
[4037.950] VM(0/0/12) play mute
[4038.050] R2(0/0/12) mute timeout
[4038.070] R2(0/0/12) Rx FW I-10: Digit 0
[4038.070] VM(0/0/12) Tx DIGIT_IND '0'
[4038.070] R2(0/0/12) Tx BW A1: Send Next Digit
22      <Call   7>      : Digit(0) at CalleeUndetermined
23      <Call   7>      : MatchedPartially
[4038.250] R2(0/0/12) MFC signal OFF, mute ON
[4038.250] VM(0/0/12) play mute
[4038.350] R2(0/0/12) mute timeout
[4038.360] R2(0/0/12) Rx FW I-10: Digit 0
[4038.360] VM(0/0/12) Tx DIGIT_IND '0'
[4038.360] R2(0/0/12) Tx BW A1: Send Next Digit
24      <Call   7>      : Digit(0) at CalleeUndetermined
25      <Call   7>      : MatchedPartially
```

```
[4038.540] R2(0/0/12) MFC signal OFF, mute ON
[4038.540] VM(0/0/12) play mute
[4038.640] R2(0/0/12) mute timeout
[4038.670] R2(0/0/12) Rx FW I-10: Digit 0
[4038.670] VM(0/0/12) Tx DIGIT_IND '0'
[4038.670] R2(0/0/12) Tx BW A1: Send Next Digit
26      <Call   7>      : Digit(0) at CalleeUndetermined
27      <Call   7>      : MatchedPerfect
[4038.670] RTA(0/0/12) Rx RCC_ADDR_CMP peerId(0/0/0)
[4038.670] R2(0/0/12) Tx BW A5: Send Calling Category and Number
[4038.850] R2(0/0/12) MFC signal OFF, mute ON
[4038.850] VM(0/0/12) play mute
[4038.950] R2(0/0/12) mute timeout
[4039.130] R2(0/0/12) MFC signal OFF, mute ON
[4039.130] VM(0/0/12) play mute
[4039.230] R2(0/0/12) mute timeout
[4039.250] R2(0/0/12) Rx FW II-1: Subscriber without priority
[4039.250] R2(0/0/12) Tx BW A5: Send Calling Category and Number
[4039.430] R2(0/0/12) MFC signal OFF, mute ON
[4039.430] VM(0/0/12) play mute
[4039.530] R2(0/0/12) Rx FW I-2: Digit 2
[4039.530] R2(0/0/12) Tx BW A5: Send Calling Category and Number
[4039.530] R2(0/0/12) mute timeout
[4039.710] R2(0/0/12) MFC signal OFF, mute ON
[4039.710] VM(0/0/12) play mute
[4039.810] R2(0/0/12) mute timeout
[4039.820] R2(0/0/12) Rx FW I-10: Digit 0
[4039.820] R2(0/0/12) Tx BW A5: Send Calling Category and Number
[4040.000] R2(0/0/12) MFC signal OFF, mute ON
[4040.000] VM(0/0/12) play mute
```

call Trace
- Analog Event –

Hook off/on

```
AP200# debug rta ipc
AP200# [2935.900] VM(0/1/0) vmOffHook
[2935.960] VM(0/1/0) vmTmoOffHook
[2935.960] VM(0/1/0) Rx OffHook
[2935.960] VM(0/1/0) Tx OFFHOOK_IND
[2935.960] VM(0/1/0) play Dial tone
[2935.960] RTA(0/1/0) Rx CC_DISCONN_REQ CZ=0, peerId(0/0/0)
[2935.960] VM(0/1/0) play Reorder tone
[2937.375] VM(0/1/0) vmOnHook
[2937.475] VM(0/1/0) vmTmoOnHook
[2937.575] VM(0/1/0) vmTmoOnHook
[2937.675] VM(0/1/0) vmTmoOnHook
[2937.675] VM(0/1/0) Rx OnHook
[2937.675] VM(0/1/0) vopp idle
[2937.675] VM(0/1/0) Tx DISCONN_CNF
```

Ring

```
AP200# debug rta ipc
AP200# [12484.065] VM(0/0/0) Rx FXO Ring Actv
[12484.065] VM(0/0/0) Tx RING_IND
[12485.075] VM(0/0/0) Rx FXO Ring Idle
[12485.075] VM(0/0/0) FXO OffHook
[12485.075] VM(0/0/0) play Dial tone
[12485.075] VM(0/0/0) Tx OFFHOOK_IND
[12491.260] VM(0/0/0) listen Reorder tone
[12491.260] VM(0/0/0) vopp idle
[12491.260] VM(0/0/0) FXO OnHook
[12491.265] VM(0/0/0) Tx DISCONN_CNF
```

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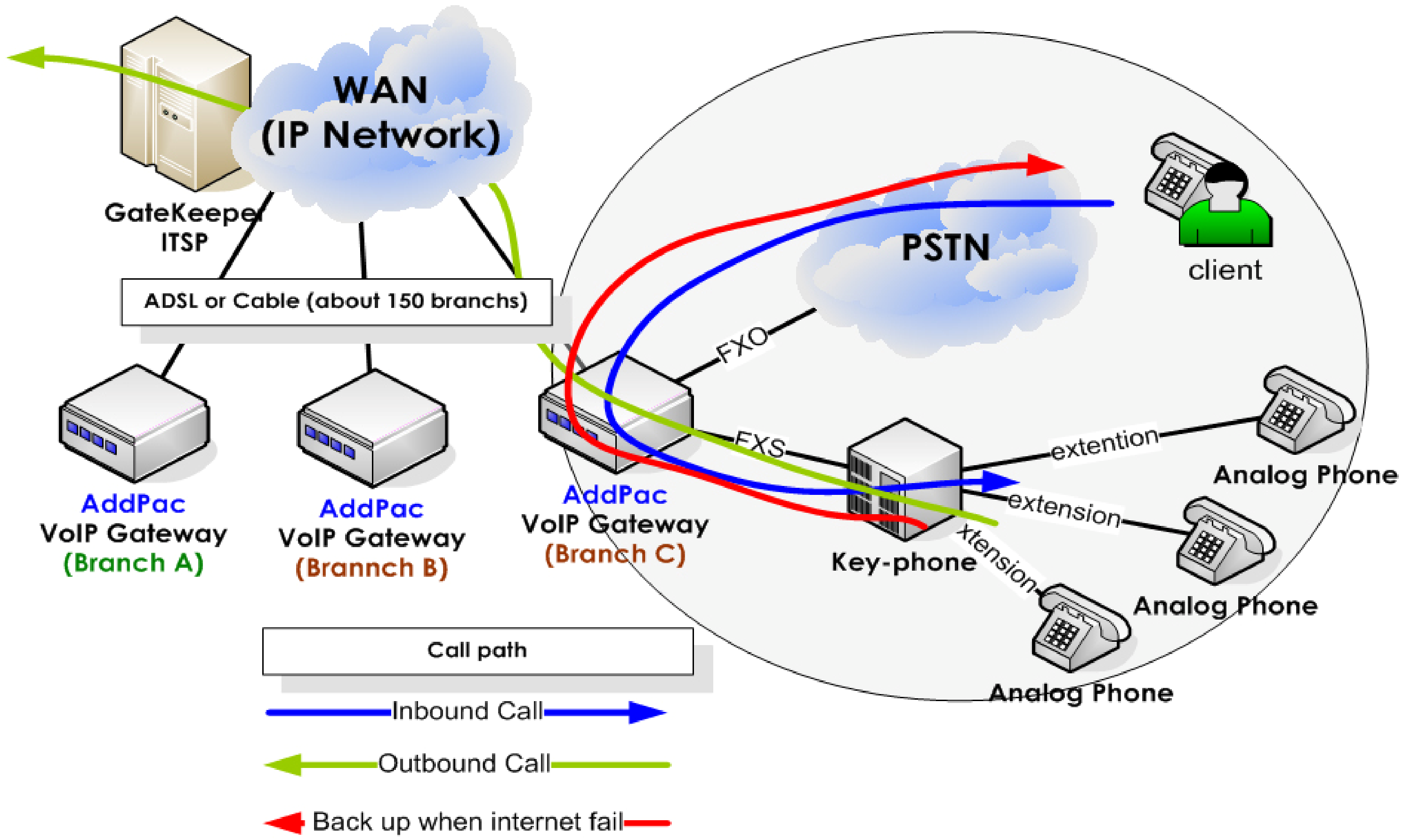
Chapter 4 Configuration Example

Configuration Example

- IP telephony Network with ITSP
- GK routed and direct call with ITSP between branch and Head Quarter

Configuration Example
- IP Telephony Network with ITSP (1/10)–



Configuration Example
- IP Telephony Network with ITSP (2/10) –

Requirement

- routing via GK Local Area/long-distance/International/Mobile Call
- Specific number which has below prefix1566 should be routed via PSTN
- 1566, 1588, 100, 106, 060, 080, 012, 015
- Inbound Call scenario should be the same with old
  redirect PSTN call(FXO) to local PABX(FXS).
- PSTN Backup if internet line has some problem.

Configuration Example
- IP Telephony Network with ITSP (3/10) –

Configuration

```
!
version 6.24
!
hostname L-ele
!
!
no ip-share enable
!
!Ethernet Interface is ADSL
interface ether0.0
 no ip address
 encapsulation pppoe
 ppp authentication pap callin
 ppp pap sent-username addpac password addpac
 ppp ipcp ms-dns
 ppp ipcp default-route
!
interface ether1.0
 no ip address
```

```
!
snmp name AP2520
!
no arp reset
!
!
!
! Voice service voip configuration.
!
voice service voip
 fax protocol t38 redundancy 0
 fax rate 9600
 h323 call start fast
 announcement element pstn-reroute
! Announcement play when Call is re-routed to
PSTN by internet fail
```

Configuration Example

- IP Telephony Network with ITSP (4/10) –

Configuration

```
!
! Voice port configuration.
!
voice-port 0/0
! FXS
!
!
voice-port 0/1
! FXS
!
!
voice-port 0/2
! FXO
 connection plar 9999
!Redirect PSTN call(FXO) to Leyphone(FXS),
 each FXS is assigned number with 9999
!
voice-port 0/3
! FXO
 connection plar 9999
!
```

```
!
voice-port 1/0
! FXS
!
!
voice-port 1/1
! FXS
!
!
voice-port 1/2
! FXO
 connection plar 9999
!
!
voice-port 1/3
! FXO
 connection plar 9999
!
```

Configuration Example

- IP Telephony Network with ITSP (5/10) –

Configuration

```
! Pots peer configuration.
!Assign e.164 number for each FXS ports.
dial-peer voice 0 pots
 destination-pattern 31082141000000
 port 0/0
!
dial-peer voice 1 pots
 destination-pattern 31082141000001
 port 0/1
!
dial-peer voice 4 pots
 destination-pattern 31082140000002
 port 1/0
!
dial-peer voice 5 pots
 destination-pattern 31082140000003
 port 1/1
!
! Set private number for connection PLAR
 PLAR is used for call redirection FXO to FXS, so e.164 9999
should not be registered.
All of FXO is same numnber 9999, so you should set « hunt-
stop » to end of hunting port which you want.
!
```

```
dial-peer voice 10 pots
 destination-pattern 9999
 port 0/0
 no register e164
!
dial-peer voice 11 pots
 destination-pattern 9999
 port 0/1
 no register e164
 preference 1
!
dial-peer voice 14 pots
 destination-pattern 9999
 port 1/0
 no register e164
 preference 2
!
dial-peer voice 15 pots
 destination-pattern 9999
 port 1/1
 no register e164
 preference 3
 huntstop
```

Configuration Example
- IP Telephony Network with ITSP (6/10) –

Configuration

```
!Prefix 060, 080 call is routed to PSTN(FXO) (Telco Provider).
dial-peer voice 42 pots
 destination-pattern 0[6-8]0.......
 port 0/2
 forward-digits last  99
!
dial-peer voice 43 pots
 destination-pattern 0[6-8]0.......
 port 0/3
 forward-digits last  99
 preference 1
!
dial-peer voice 46 pots
 destination-pattern 0[6-8]0.......
 port 1/2
 forward-digits last  99
 preference 2
!
dial-peer voice 47 pots
 destination-pattern 0[6-8]0.......
 port 1/3
 forward-digits last  99
 preference 3
 huntstop
```

```
! Prefix 1566,1588 call is routed to PSTN(FXO) (Telco Provider)..
dial-peer voice 22 pots
 destination-pattern 1[5-6][68][68]....
 port 0/2
 forward-digits last  99
!
dial-peer voice 23 pots
 destination-pattern 1[5-6][68][68]....
 port 0/3
 forward-digits last  99
 preference 1
!
dial-peer voice 26 pots
 destination-pattern 1[5-6][68][68]....
 port 1/2
 forward-digits last  99
 preference 2
!
dial-peer voice 27 pots
 destination-pattern 1[5-6][68][68]....
 port 1/3
 forward-digits last  99
 preference 3
 huntstop
```

Configuration Example

- IP Telephony Network with ITSP (7/10) –

Configuration

```
!10X,11X number is routed to PSTN(FXO) (Telco Provider).
dial-peer voice 32 pots
 destination-pattern 1[0-1].
 port 0/2
 forward-digits last  99
!
dial-peer voice 33 pots
 destination-pattern 1[0-1].
 port 0/3
 forward-digits last  99
 preference 1
!
dial-peer voice 36 pots
 destination-pattern 1[0-1].
 port 1/2
 forward-digits last  99
 preference 2
!
dial-peer voice 37 pots
 destination-pattern 1[0-1].
 port 1/3
 forward-digits last  99
 preference 3
 huntstop
```

```
!the configuration Hunting for PSTN Backup.
dial-peer voice 102 pots
 destination-pattern T
 port 0/2
 preference 1
!
dial-peer voice 103 pots
 destination-pattern T
 port 0/3
 preference 2
!
dial-peer voice 106 pots
 destination-pattern T
 port 1/2
 preference 3
!
dial-peer voice 107 pots
 destination-pattern T
 port 1/3
 preference 4
 huntstop
```

Configuration Example

- IP Telephony Network with ITSP (8/10) –

Configuration

```
!making phone-book(number-ring plan) for all of number .
dial-peer voice 1000 voip
 destination-pattern T
 session target ras
 dtmf-relay h245-alphanumeric
 translate-outgoing called-number 0
!
dial-peer voice 1010 voip (local area)
 destination-pattern 0[3-6][1-5][2-9]......
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1011 voip (local area)
 destination-pattern 0[3-6][1-5]1.T
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1020 voip
 destination-pattern 1[2-9].T
 session target ras
 dtmf-relay h245-alphanumeric
 translate-outgoing called-number 0
```

```
dial-peer voice 1021 voip
 destination-pattern 2.......
 session target ras
 dtmf-relay h245-alphanumeric
 translate-outgoing called-number 0
!
dial-peer voice 1022 voip
 destination-pattern [45789]......
 session target ras
 dtmf-relay h245-alphanumeric
 translate-outgoing called-number 0
!
dial-peer voice 1023 voip
 destination-pattern [36].....T
 session target ras
 dtmf-relay h245-alphanumeric
 translate-outgoing called-number 0
!
dial-peer voice 1030 voip (mobile)
 destination-pattern 011[19].......
 session target ras
 dtmf-relay h245-alphanumeric
!
```

Configuration Example

- IP Telephony Network with ITSP (9/10) –

Configuration

```
dial-peer voice 1031 voip (mobile)
 destination-pattern 011[2-8]......
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1032 voip (mobile)
 destination-pattern 01[6-9]9.......
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1033 voip (mobile)
 destination-pattern 01[6-9][2-8]......
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1034 voip (114 service by SK Telecom)
 destination-pattern 01[17]114F
 session target ras
 dtmf-relay h245-alphanumeric
 !
```

```
!
dial-peer voice 1035 voip (114 Service by KTF,LG Telecom)
 destination-pattern 01[689]114
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1040 voip
 destination-pattern 082.T
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1041 voip
 destination-pattern 0808[5-6].....
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1042 voip
 destination-pattern 060.......
 session target ras
 dtmf-relay h245-alphanumeric
!
```

Configuration Example
- IP Telephony Network with ITSP (10/10) –

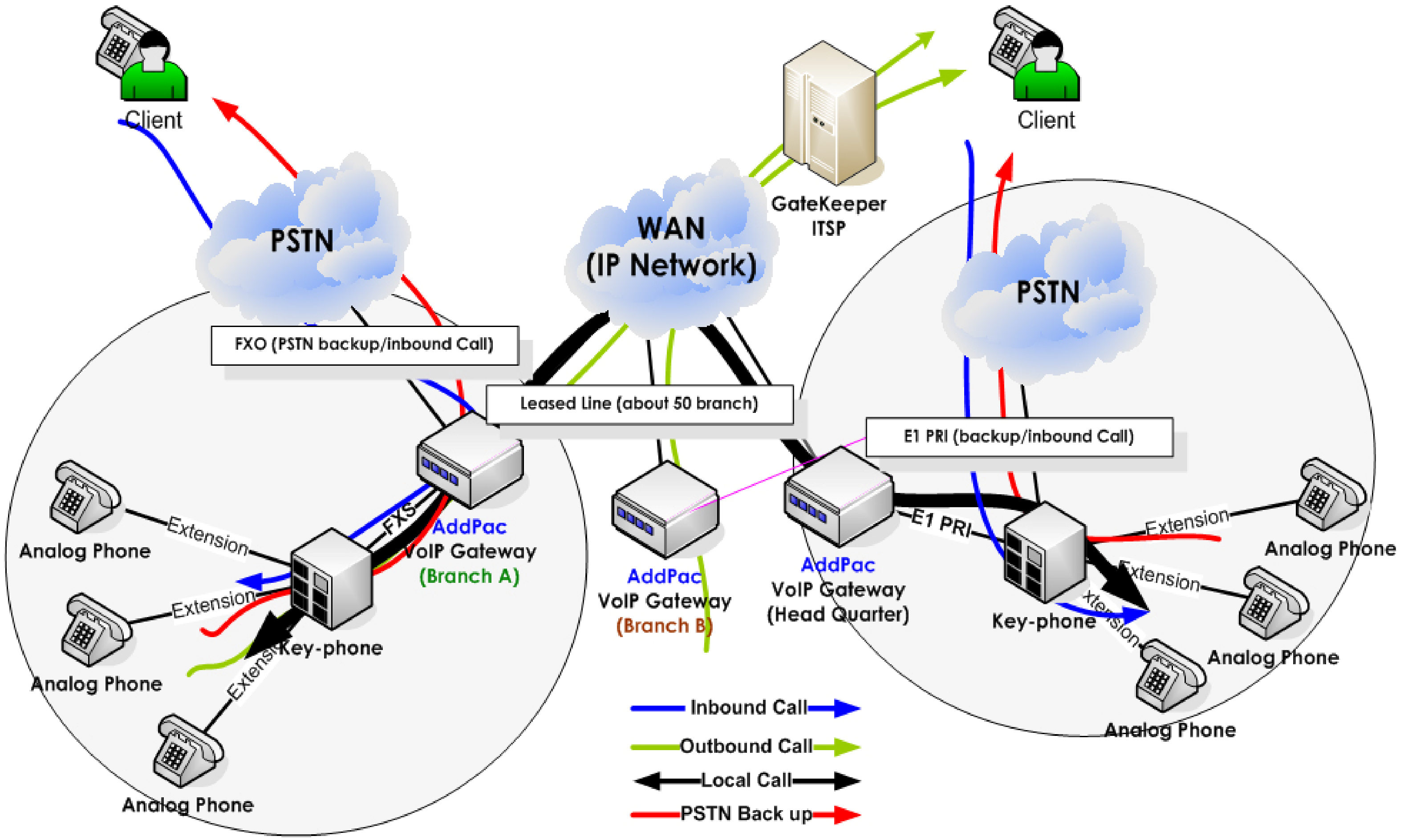
Configuration

```
!
dial-peer voice 1043 voip
 destination-pattern 0505.......
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1044 voip
 destination-pattern 0303[25].......
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1045 voip
 destination-pattern 1544....
 session target ras
 dtmf-relay h245-alphanumeric
!
dial-peer voice 1050 voip (International)
 destination-pattern 00.........T
 session target ras
 dtmf-relay h245-alphanumeric
!
```

```
! Gateway configuration.
!
gateway
 h323-id 12345678  (H323-ID)
 gkip 1.2.3.4   (GK ip)
 register   (register to GK)
!
!
! Translation Rule configuration.
!
translation-rule 0
 rule 0      [1-9]                 02%01%99
 (forlocal area code adding to destination number  - Number
Translation )
!
```

Configuration Example

- GK routed and direct call with ITSP between branch and head Quarter (1/6) –



Configuration Example

- GK routed and direct call with ITSP between branch and head Quarter (2/6) –

Requirement

- Direct Internal call not through GK
  for this, assign virtual area code to each branch
- routing via GK Local Area/long-distance/International/Mobile Call
- PSTN back-up when VoIP network fail
  Head Quarter
    PSTN backup by E1 link down using PABX
  Branch
    PSTN backup by FXO hunting(directly connected telco line with FXO)
    enable Announcement to notify user.
- Inbound call
  Head Quarter is the same with old
  Branch is redirected by AddPac gateway (PSTN(FXO) to FXS re-routing)

Configuration Example

- GK routed and direct call with ITSP between branch and head Quarter (3/6) –

Configuration (HQ)

```
! PRI controller configuration.
! ! ISDN PRI Configuration.
controller e1 1/0
 channel-group timeslots 1-31  0
 isdn protocol-emulate user
!
! Voice service voip configuration.
! !VoIP related configuration.
voice service voip
 fax protocol t38 redundancy 0
 fax rate 9600
 h323 call start fast
 timeout tttl 600
 timeout tterm 1800
! !set PSTN backup condition.
 busyout monitor gatekeeper
 busyout monitor voip-interface
!
```

```
! !Set Action when busyout status-> action = E1 interface Link down.
!
voice-port 1/0 0
! E1
 busyout action port-down
!
!
!
!
```

Configuration Example

- GK routed and direct call with ITSP between branch and head Quarter (4/6) –

Configuration (HQ)

```
! Pots peer configuration.
! ! ! ! Set e.164 number for registration to GK.
dial-peer voice 1 pots
 destination-pattern 12345678
 port 1/0  0
 preference 9
 huntstop
! ! ! ! ! set virtual area code(local) for internal call branch to branch .
!
dial-peer voice 11 pots
 destination-pattern *10T
 port 1/0  0
 huntstop
!
dial-peer voice 21 pots
 destination-pattern [0-9].T
 port 1/0  0
 translate-outgoing called-number 2001
 preference 1
 huntstop
!
```

```
!
dial-peer voice 22 pots
 destination-pattern *1[1-9]T
 port 1/0  0
 translate-outgoing called-number 2001
 preference 1
 huntstop
!
dial-peer voice 23 pots
 destination-pattern *2[0-9]T
 port 1/0  0
 translate-outgoing called-number 2001
 preference 1
 huntstop
!
……
………
…………….
```

Configuration Example

- GK routed and direct call with ITSP between branch and head Quarter (5/6) –

Configuration (HQ)

```
!
! Voip peer configuration.
! ! set dial-number plan for through GK(RAS)
!
dial-peer voice 1001 voip
 destination-pattern [0-9].T
 session target ras
 dtmf-relay h245-alphanumeric
 translate-outgoing called-number 1001
 huntstop
!
dial-peer voice 1002 voip
 destination-pattern 01.........
 session target ras
 dtmf-relay h245-alphanumeric
 huntstop
!
```

```
! ! set virtual area code(remote) for internal call branch to branch .
!
dial-peer voice 2011 voip
 destination-pattern *11T
 session target 1.0.0.1
 dtmf-relay h245-alphanumeric
 translate-outgoing calling-number 2000
!
dial-peer voice 2012 voip
 destination-pattern *12T
 session target 1.0.0.2
 dtmf-relay h245-alphanumeric
 translate-outgoing calling-number 2000
!
......
........
.........
```

Configuration Example

- GK routed and direct call with ITSP between branch and head Quarter (6/6) –

Configuration (HQ)

```
! ! ! GK related parameter
!

! Gateway configuration.
!
gateway
 h323-id addpac
 gkip 1.2.3.4
 register
```

```
! ! Number Translation Rule
!
translation-rule 2000
 rule 0      T                   *10%99
!
translation-rule 2001
 rule 0      [0-9].T              70%01%99
 rule 10     *10T                 7030161%04%99
 rule 11     *11T                 7003232618%04%99
 rule 12     *12T                 7003190691%04%99
```

Configuration (Branch)

Skip