

# VoIP Gateway/IP-PBX Interworking with Skype



IPNext320  
Hybrid IP-PBX



IPNext50  
IP-PBX



IPNext20  
IP-PBX



AP100B  
VoIP Gateway

**AddPac**

**AddPac Technology**

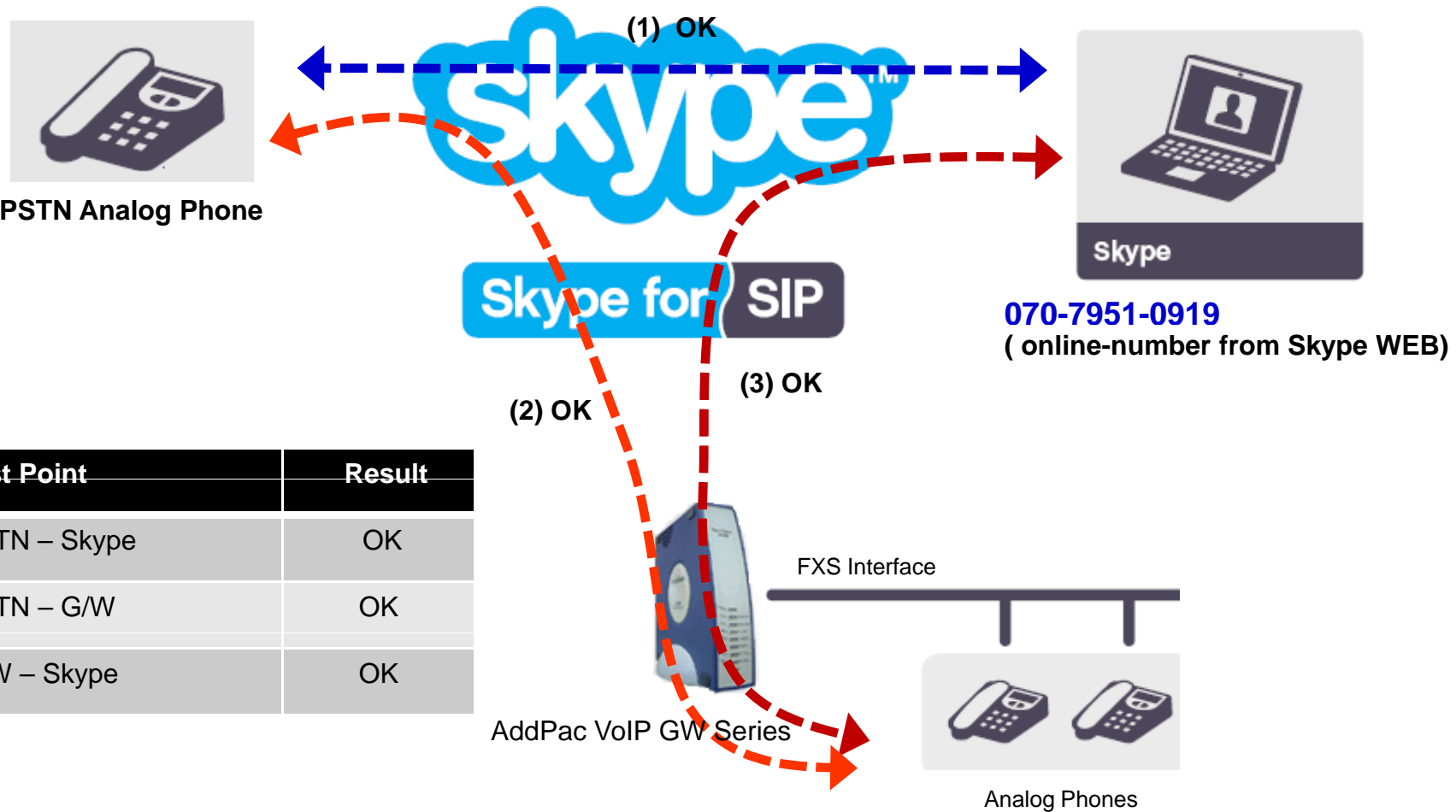
2012, Sales and Marketing

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# Skype Interworking Test (GW – Skype :using online-number)

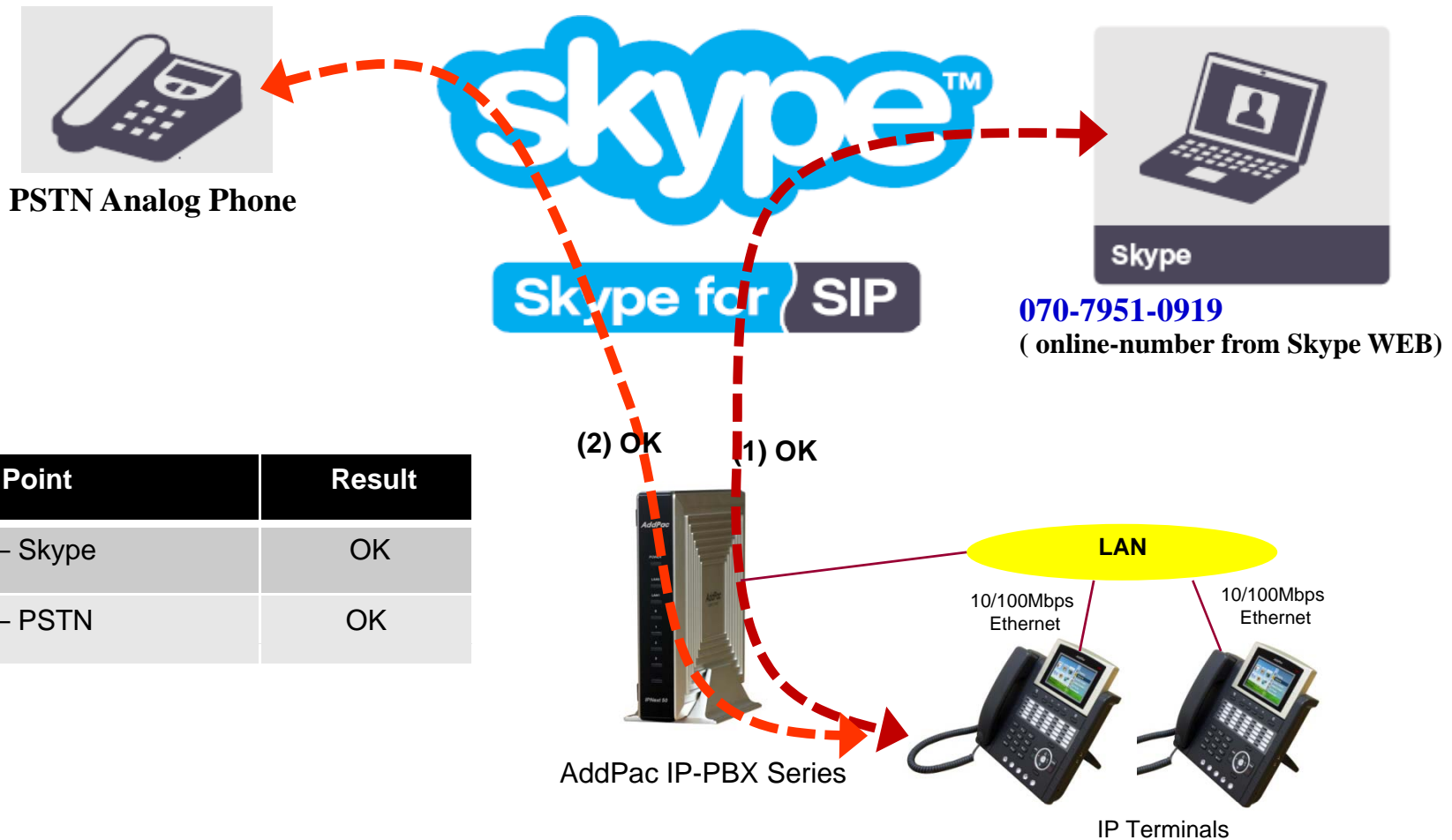
## Test System Diagram (GW – Skype)



Test Point	Result
PSTN – Skype	OK
PSTN – G/W	OK
G/W – Skype	OK

# Skype Interworking Test (IP-PBX – Skype :using online-number)

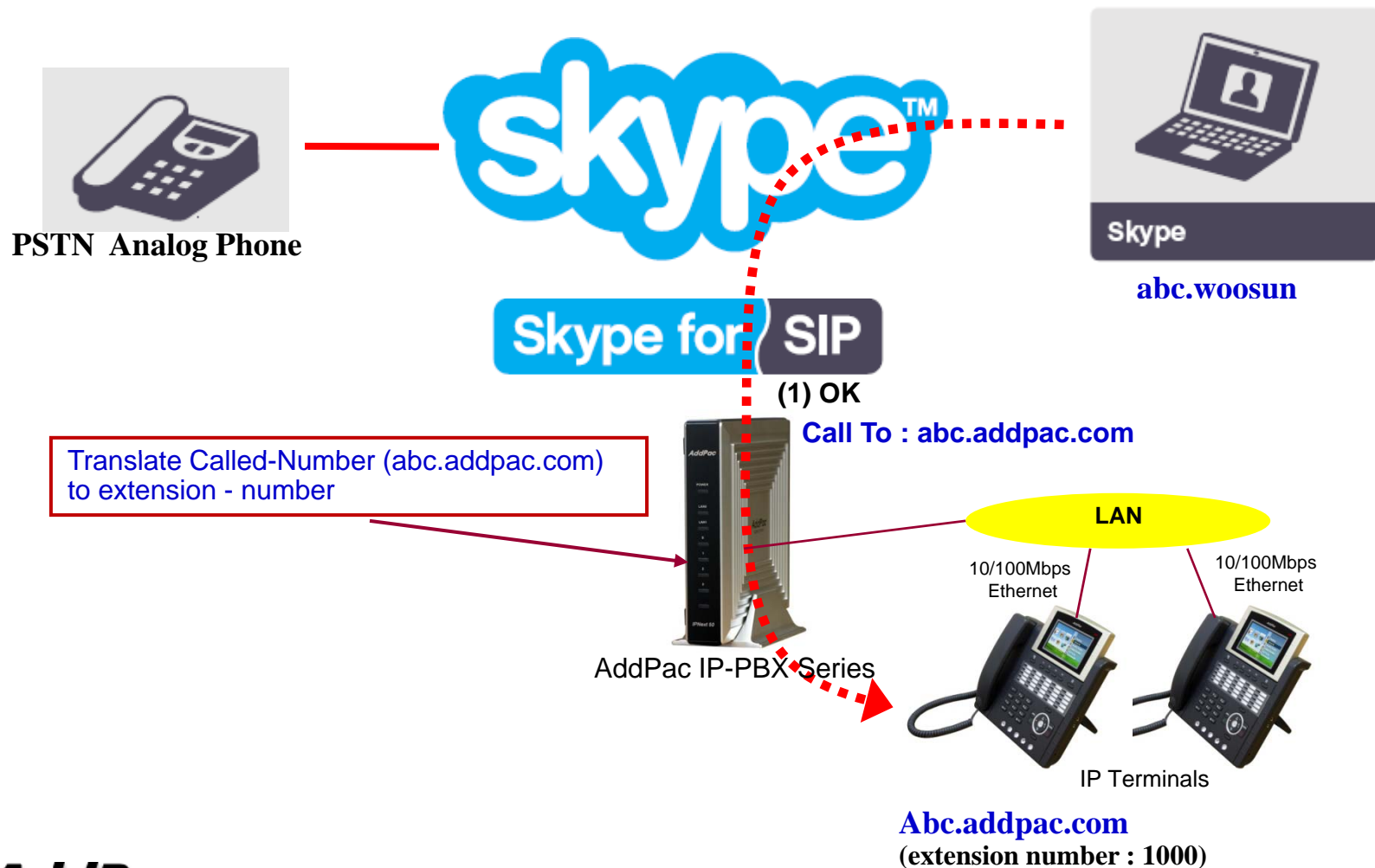
## Test System Diagram (IP-PBX – Skype)



Test Point	Result
PBX– Skype	OK
PBX– PSTN	OK

# Skype Interworking Test (IP-PBX – Skype :using **Skype-name**)

## Test System Diagram (IP-PBX – Skype)



# Skype Configuration for IP-PBX or G/W (1/3)

## Skype for SIP Beta

The screenshot displays the 'Skype for SIP Beta Profile' page. The left sidebar contains navigation links: Account details, Purchase Skype credit, Add members, Redeem voucher, Manage online numbers, Group people, Order list, Allocation report, Payment preferences, and Skype for SIP Beta (highlighted). The main content area includes a 'Business Control Panel' header with 'Help' and 'Sign out' links, and a balance of €5,10. Below this are tabs for 'Profile overview', 'Calling', 'Online numbers', and 'Caller ID'. The 'SIP authentication' section explains that registration is used and provides the following information:

Registration (username and password) <input checked="" type="radio"/>		IP address <input type="radio"/>	
You will need this information to configure your PBX			
SIP User:	99051000003457		
Password:	e8kVVTU2		
Skype for SIP domain	sip.skype.com		
UDP Port:	5060		
SIP user successfully registered at sip.skype.com Last registration: March 29, 2010 at 01:39 GMT			
<a href="#">Generate a new password</a>			

A red box highlights the registration information, with a list of required fields: user, password, SIP domain, and UDP port. To the right of the box are images of a black and a blue VoIP gateway device.

# Skype Configuration for IP-PBX or G/W (2/3)

## Manage online numbers

skype AddPac Technology [Business Control Panel](#) | [Help](#) | [Sign out](#)

People Company €5,10 Add 2 people Add

- [Account details](#)
- [Purchase Skype credit](#)
- [Add members](#)
- [Redeem voucher](#)
- Manage online numbers**
- [Group people](#)
- [Order list](#)
- [Allocation report](#)
- [Payment preferences](#)
- [Skype for SIP Beta](#)

### Manage online numbers

1 online number(s) available [Purchase more online numbers](#)

Select all	Assigned to	Expires
<input type="checkbox"/>	<a href="#">SIP profile: Headquarter</a>	Jun 24, 2010



Extend selected numbers (3 months)

[Assign and reassign your online numbers here.](#)

Assign the online-number for receiving a PSTN or VoIP call.

# Skype Configuration for IP-PBX or G/W (3/3)

## Configure Extension Line

People Company  €5,10 [Add](#)  2 people [Add](#)

[Account details](#)  
[Purchase Skype credit](#)  
[Add members](#)  
[Redeem voucher](#)  
[Manage online numbers](#)  
[Group people](#)  
[Order list](#)  
[Allocation report](#)  
[Payment preferences](#)  
**Skype for SIP Beta**

### Skype for SIP Beta Profile

[Profile overview](#) **Calling** [Online numbers](#) [Caller ID](#) [« Back to profile list](#)

#### Headquarter - Calling

Business accounts can have their incoming calls redirected to this profile.  
[Create more business accounts.](#)

#### Business Accounts

Change or add business accounts to this profile.

Business account	Extension number	
wshwang.addpac.com	<u>1002</u>	<a href="#">Update</a> <a href="#">Remove</a>

Assign the extension line number for receiving a PSTN or VoIP call.






# Configuration for Skype Application

## Online number for Skype Application

You have

- Free calls and video
- Call phones
- Online number**
- Skype To Go number
- Voicemail
- Send SMS
- Call forwarding
- Caller ID

Your online numbers

 **(70) 7951-0919**  
Korea, South · Expires on June 22, 2010 · [Extend](#)

**Need more online numbers?**  
Buy more online numbers so even more friends can save money when calling you.

[Get another online number](#)

Assign the online-number for receiving a PSTN or VoIP call.



# AddPac VoIP Gateway Configuration

## AddPac G/W configuration

```
dial-peer voice 0 pots
destination-pattern 827078931524
port 0/0
no register e164
user-name 99051000003457
user-password e8kVVTU2Jxxxxx
!
dial-peer voice 1 pots
destination-pattern 99051000003457
port 0/0
user-password e8kVVTU2Jxxxxx

sip-ua
sip-server sip.skype.com
called-party-number to-field
```

**Configure online-number for receiving a call.  
Also, The authentication information should be configured.**  
*- reference page 4-5*

**The authentication information should be configured for REGISTRATION.**  
*- reference page 5*

**Configure SIP-Proxy Server information.  
The option(called-party-number) should be configured for extracting called-number from 'To filed'.**  
*- reference page 5*

# AddPac IP-PBX Configuration(1/4)

## Skype Proxy Server Configuration

**General** Routing Pattern Phone Number Call Control Options

Proxy Server Name: Skype  
Description:   
Device Pool: default [Edit]  
Location: N/A [Select]  
Security Profile: <N/A> [Edit]

SIP User Name: 99051000003457  
SIP Password:   
Local Domain: public  
Rtp-2833  
UDP

out: 60 (10-86400 sec)

RTP Proxy Required  Use Music On Hold  
 Use Local Hostname at Registered Domain Name  Nortel Hold Method  
 Use Username at Registered User Information  REFER Method Supported  
 Register

Ok Cancel

No.	Address	Port
1	sip.skype.com	5060

**The authentication information should be configured.**  
*- reference the page 4*

**Configure SIP-Proxy Server information.**  
*- reference the page 5*

# AddPac IP-PBX Configuration(2/4)

## Translation Rule Configuration

Number Translation Rule				
No	Number Translation Rule	Input Matched Pattern	Substituted Pattern	Description
1	외부발신_called	9T	%02%99	
		8T	%02%99	
		53T	%03%99	
		54T	%03%99	
		55T	%03%99	
		56T	%03%99	
		57T	%03%99	
2	Skype_outbound_called	070T	82%02%99	
3	Skype_inbound_called	8270T	0%03%99	

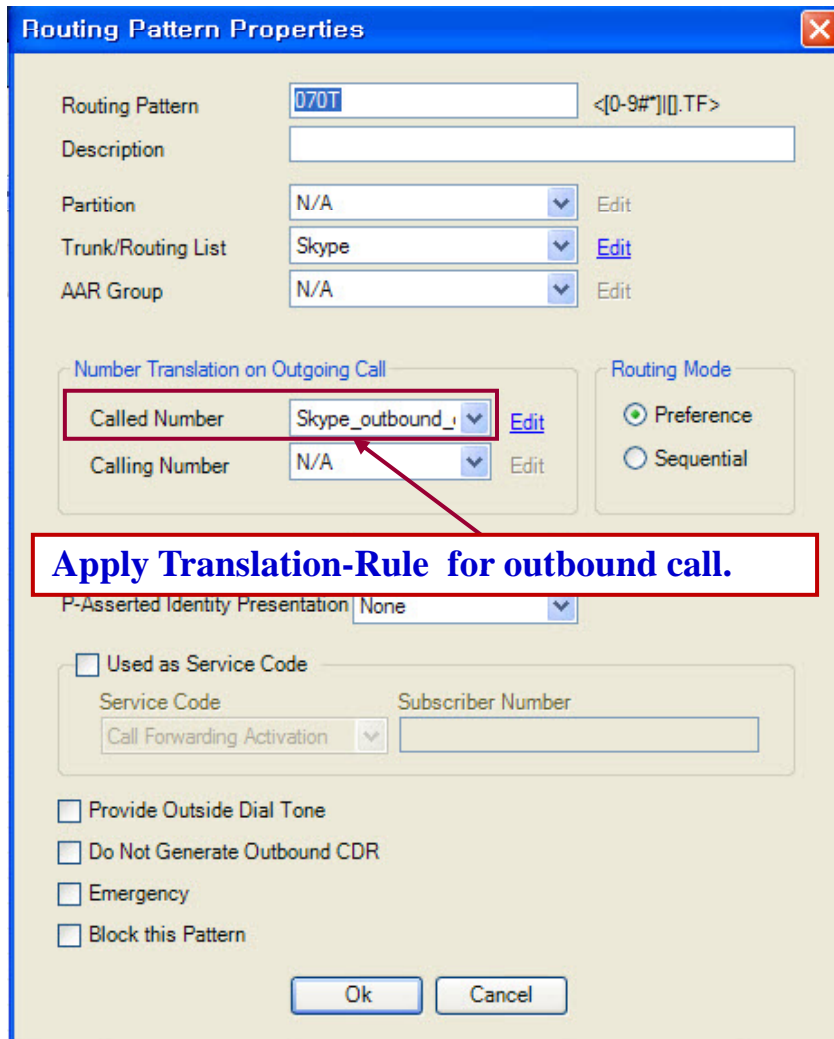
**Configure Translation-Rules for inbound and outbound call.**

**For the outbound call starting with '070', Eliminate one digit, and then insert '82' digits.  
( ex: called-number 070-8888-9999 → 8270-8888-9999 )**

**For the inbound call starting with '8270', Eliminate two digits, and then insert '0' digit.  
( ex: called-number 8270-8888-9999 → 070-8888-9999 )**

# AddPac IP-PBX Configuration(3/4)

## Apply Translation Rule



**Routing Pattern Properties**

Routing Pattern: 070T <[0-9#\*][].TF>  
Description:

Partition: N/A Edit  
Trunk/Routing List: Skype Edit  
AAR Group: N/A Edit

Number Translation on Outgoing Call

Called Number: Skype\_outbound\_1 Edit  
Calling Number: N/A Edit

Routing Mode  
 Preference  
 Sequential

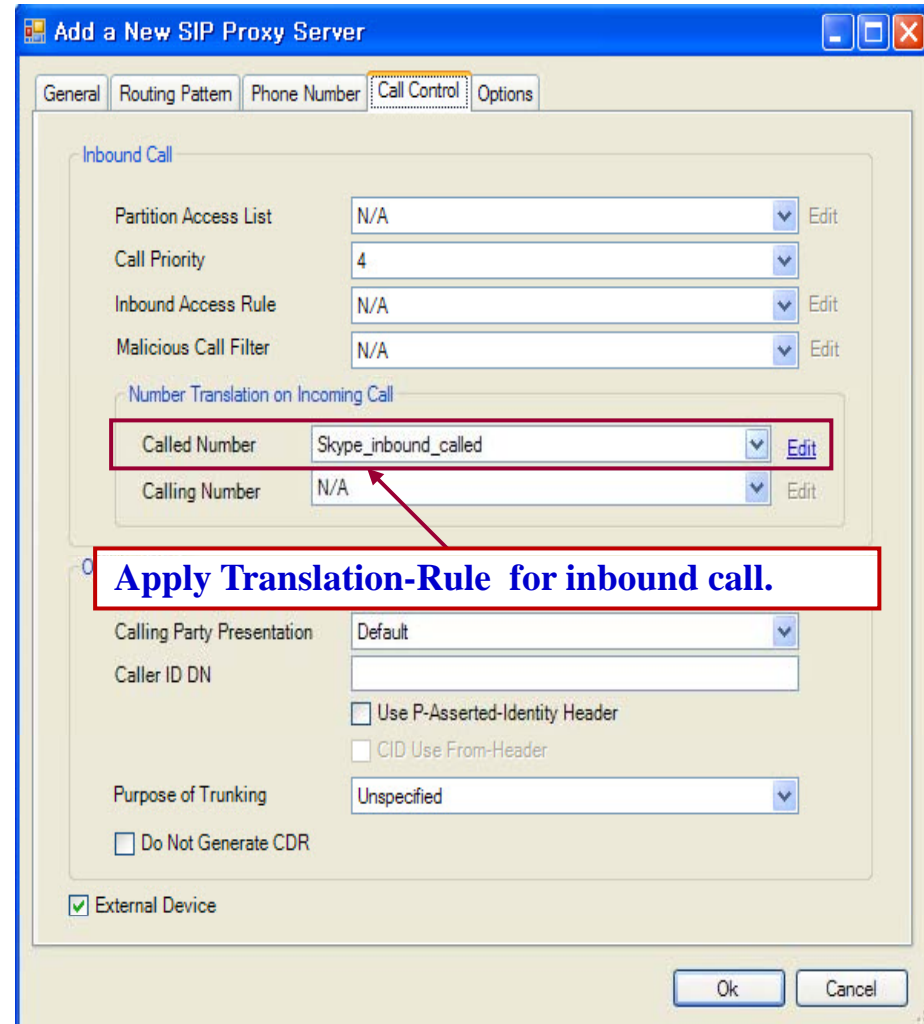
P-Asserted Identity Presentation: None

Used as Service Code  
Service Code:  Subscriber Number:   
Call Forwarding Activation:

Provide Outside Dial Tone  
 Do Not Generate Outbound CDR  
 Emergency  
 Block this Pattern

Ok Cancel

**Apply Translation-Rule for outbound call.**



**Add a New SIP Proxy Server**

General Routing Pattern Phone Number Call Control Options

Inbound Call

Partition Access List: N/A Edit  
Call Priority: 4  
Inbound Access Rule: N/A Edit  
Malicious Call Filter: N/A Edit

Number Translation on Incoming Call

Called Number: Skype\_inbound\_called Edit  
Calling Number: N/A Edit

Calling Party Presentation: Default  
Caller ID DN:   
 Use P-Asserted-Identity Header  
 CID Use From-Header

Purpose of Trunking: Unspecified  
 Do Not Generate CDR

External Device

Ok Cancel

**Apply Translation-Rule for inbound call.**

# AddPac IP-PBX Configuration(4/4)

## Configure Routing Pattern

The image shows two overlapping configuration windows. The background window is 'Routing Pattern Properties' with the following fields:

- Routing Pattern: 070T <[0-9#\*]||.TF>
- Description: (empty)
- Partition: N/A
- Trunk/Routing List: Skype
- AAR Group: N/A
- Number Translation on Outgoing Call: (checked)
- Called Number: Skype\_outbound\_1
- Calling Number: Skype\_outbound\_1
- Display Name Presentation: None
- P-Asserted Identity Presentation: None
- Used as Service Code: (unchecked)
- Service Code: Call Forwarding Activation
- Subscriber Number: (empty)
- Provide Outside Dial Tone: (unchecked)
- Do Not Generate Outbound CDR: (unchecked)
- Emergency: (unchecked)
- Block this Pattern: (unchecked)

The foreground window is 'Number Translation Properties' with the following fields:

- Name: Skype\_outbound\_calling
- Description: (empty)
- Number Translation Rules table:

No	Input Matched Pattern	Substituted Pattern
1	T	99051000003457%98

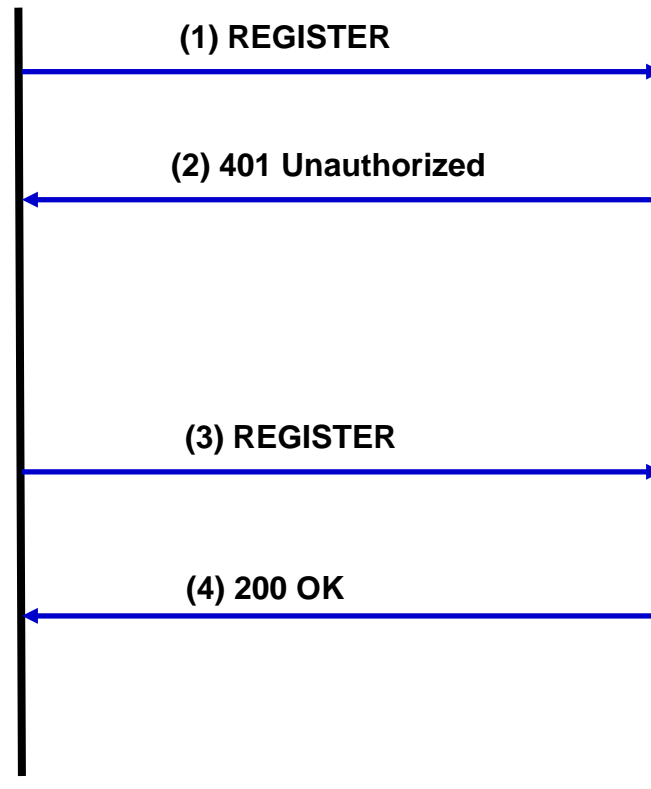
Red boxes highlight the 'Number Translation on Outgoing Call' section in the Routing Pattern Properties window, the 'Calling Number' field, and the first row of the Number Translation Rules table. A red arrow points from the 'Calling Number' field to the 'Substituted Pattern' cell. A red box at the bottom contains the following text:

**Configure the calling number translation rule.  
( Skype Proxy Server allows only registered user ID.)  
- reference the page 5**

# SIP REGISTER (1/3)



AddPac IP-PBX Series



# SIP REGISTER (2/3)

## (1) REGISTER

```
Request-Line: REGISTER sip:sip.skype.com SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk494b7346a41
From: <sip:99051000003457@sip.skype.com>;tag=494b7346a4
To: sip:99051000003457@sip.skype.com
Call-ID: 495fb04b-9947-7314-8046-0002a4ff4869@60.196.6.77
CSeq: 1 REGISTER
Date: Mon, 29 Mar 2010 17:05:29 GMT
User-Agent: AddPac SIP Gateway
Contact: <sip:99051000003457@60.196.6.77>;expires=60
Expires: 60
Content-Length: 0
Max-Forwards: 70
```

**SIP user is inserted to From-Field.**  
*- reference the page 5*

## (2) 401 Unauthorized

```
Status-Line: SIP/2.0 401 unauthorized
Status-Code: 401
[Resent Packet: False]
Message Header
From: <sip:99051000003457@sip.skype.com>;tag=494b7346a4
To: <sip:99051000003457@sip.skype.com>;tag=05aed4eb43523e287156e2da6464d890.fe30
Call-ID: 495fb04b-9947-7314-8046-0002a4ff4869@60.196.6.77
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk494b7346a41
www-Authenticate: Digest realm="sip.skype.com", nonce="4bb05f6b000128eb7223ee8f101719a27202e08df27d98d7", algorithm=MD5
Server: OpenSIPS
Content-Length: 0
```



# SIP REGISTER (3/3)

## (3) REGISTER

```
Request-Line: REGISTER sip:sip.skype.com SIP/2.0
  Method: REGISTER
  [Resent Packet: False]
Message Header
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk494b7346a42
  From: <sip:99051000003457@sip.skype.com>;tag=494b7346a4
  To: sip:99051000003457@sip.skype.com
  Call-ID: 495fb04b-9947-7314-8046-0002a4ff4869@60.196.6.77
  CSeq: 2 REGISTER
  Date: Mon, 29 Mar 2010 17:05:29 GMT
  User-Agent: AddPac SIP Gateway
  Authorization: Digest username="99051000003457", realm="sip.skype.com", nonce="4bb05f6b000128eb7223ee8f101719a27202e08df27d98d7", uri="sip:sip.skype.com", response="9e7434a787cdef395518b"
  Contact: <sip:99051000003457@60.196.6.77>;expires=60
  Expires: 60
  Content-Length: 0
  Max-Forwards: 70
```

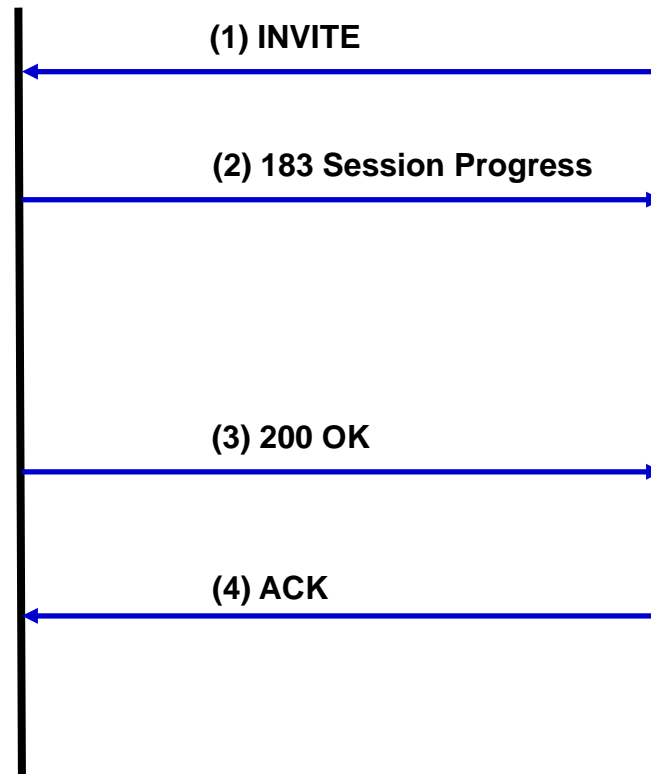
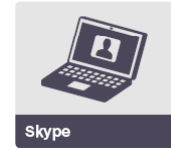
## (4) 200 OK

```
Status-Line: SIP/2.0 200 OK
  Status-Code: 200
  [Resent Packet: False]
Message Header
  From: <sip:99051000003457@sip.skype.com>;tag=494b7346a4
  To: <sip:99051000003457@sip.skype.com>;tag=05aed4eb43523e287156e2da6464d890.d621
  Call-ID: 495fb04b-9947-7314-8046-0002a4ff4869@60.196.6.77
  CSeq: 2 REGISTER
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk494b7346a42
  Contact: <sip:99051000003457@60.196.6.77>;expires=60
  Server: OpenSIPS
  Expires: 60
  Content-Length: 0
```

# Inbound Call from Skype (1/5)



AddPac IP-PBX Series



# Inbound Call from Skype (2/5)

## (1) INVITE

```
⊕ Request-Line: INVITE sip:99051000003457@60.196.6.77 SIP/2.0
⊖ Message Header
⊕ From: <sip:Anonymous@sip.skype.com>;tag=a4a109cc-13c4-4bb0601c-2729e3f7-36880b6f
⊕ To: <sip:827078931524@sip.skype.com>
  Call-ID: CXC-59-6942a370-a4a109cc-13c4-4bb0601c-2729e3f7-2afee549
  CSeq: 1 INVITE
  Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hg4bk-287ea-4bb0601c-2729e3f7-3242e07c
  Max-Forwards: 12
  User-Agent: sipgw-1.0
  Privacy: id
  P-Asserted-Identity: <sip:Anonymous@sip.skype.com>
  Remote-Party-ID: <sip:Anonymous@sip.skype.com>;party=calling;screen=yes;privacy=full
  Allow: INVITE,ACK,CANCEL,OPTIONS,BYE
⊕ Contact: <sip:Anonymous@204.9.161.164:5060;transport=udp>
  Content-Type: application/sdp
  Content-Length: 263
⊖ Message body
⊖ Session Description Protocol
  Session Description Protocol Version (v): 0
⊕ Owner/Creator, Session Id (o): Anonymous 1269850140 1269850140 IN IP4 204.9.161.164
  Session Name (s): Skype call
⊕ Connection Information (c): IN IP4 204.9.161.164
⊕ Time Description, active time (t): 0 0
⊕ Media Description, name and address (m): audio 28924 RTP/AVP 18 0 8 101
⊕ Media Attribute (a): rtpmap:18 G729/8000
⊕ Media Attribute (a): rtpmap:0 PCMU/8000
⊕ Media Attribute (a): rtpmap:8 PCMA/8000
⊕ Media Attribute (a): rtpmap:101 telephone-event/8000
⊕ Media Attribute (a): fmp:18 annex=no
```

# Inbound Call from Skype (3/5)

## (2) 183 Session Progress

```
⊕ Status-Line: SIP/2.0 183 Session Progress
⊖ Message Header
  Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hg4bk-287ea-4bb0601c-2729e3f7-3242e07c
  ⊕ From: <sip:Anonymous@sip.skype.com>;tag=a4a109cc-13c4-4bb0601c-2729e3f7-36880b6f
  ⊕ To: <sip:827078931524@sip.skype.com>;tag=194b5b49a4
  Call-ID: CXC-59-6942a370-a4a109cc-13c4-4bb0601c-2729e3f7-2afee549
  CSeq: 1 INVITE
  User-Agent: AddPac SIP Gateway
  ⊕ Contact: <sip:99051000003457@60.196.6.77>
  Content-Type: application/sdp
  Content-Length: 177
⊖ Message body
  ⊖ Session Description Protocol
    Session Description Protocol version (v): 0
    ⊕ Owner/Creator, Session Id (o): addpac 1269850137 1269850137 IN IP4 60.196.6.77
    Session Name (s): AddPac Gateway SDP
    ⊕ Connection Information (c): IN IP4 60.196.6.77
    ⊕ Time Description, active time (t): 1269850137 0
    Session Attribute (a): sendonly
    ⊕ Media Description, name and address (m): audio 26128 RTP/AVP 18
    ⊕ Media Attribute (a): rtpmap:18 G729/8000
```

# Inbound Call from Skype (4/5)

## (3) 200 OK

```
⊕ Status-Line: SIP/2.0 200 OK
⊖ Message Header
  Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hg4bk-287ea-4bb0601c-2729e3f7-3242e07c
  ⊕ From: <sip:Anonymous@sip.skype.com>;tag=a4a109cc-13c4-4bb0601c-2729e3f7-36880b6f
  ⊕ To: <sip:827078931524@sip.skype.com>;tag=194b5b49a4
  Call-ID: CXC-59-6942a370-a4a109cc-13c4-4bb0601c-2729e3f7-2afee549
  CSeq: 1 INVITE
  User-Agent: AddPac SIP Gateway
  ⊕ Contact: <sip:99051000003457@60.196.6.77>
  Content-Type: application/sdp
  Content-Length: 248
⊖ Message body
  ⊖ Session Description Protocol
    Session Description Protocol version (v): 0
    ⊕ Owner/Creator, Session Id (o): 07078931524 1269882543 1269882543 IN IP4 172.17.111.211
    Session Name (s): AddPac Gateway SDP
    ⊕ Connection Information (c): IN IP4 172.17.111.211
    ⊕ Time Description, active time (t): 1269882543 0
    ⊕ Media Description, name and address (m): audio 23106 RTP/AVP 18 101
    ⊕ Media Attribute (a):ptime:20
    ⊕ Media Attribute (a):rtptime:18 G729/8000/1
    ⊕ Media Attribute (a):rtptime:101 telephone-event/8000/1
    ⊕ Media Attribute (a):fmtp:101 0-15
```

# Inbound Call from Skype (5/5)

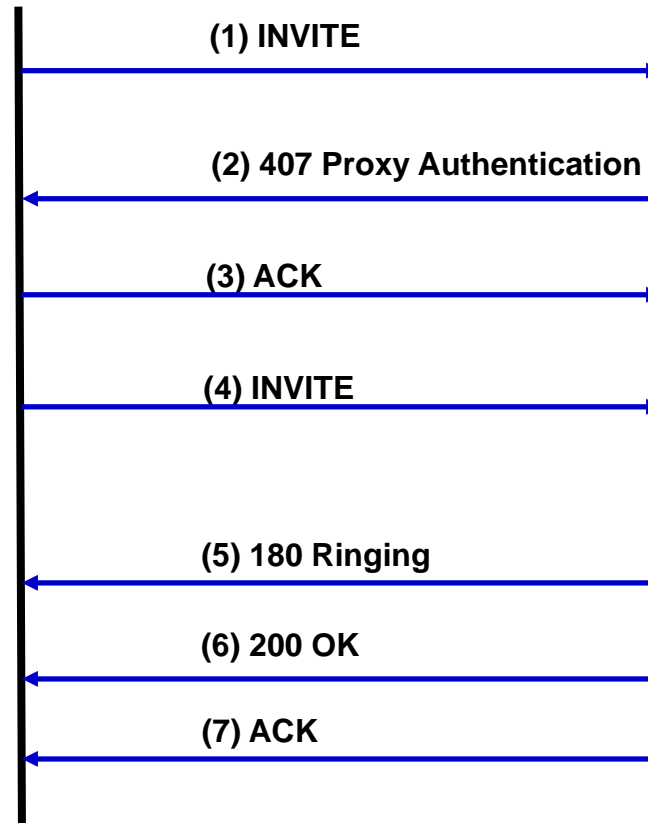
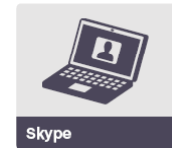
## (4) ACK

```
⊕ Request-Line: ACK sip:99051000003457@60.196.6.77 SIP/2.0
⊖ Message Header
⊕ From: <sip:Anonymous@sip.skype.com>;tag=a4a109cc-13c4-4bb0601c-2729e3f7-36880b6f
⊕ To: <sip:827078931524@sip.skype.com>;tag=194b5b49a4
  Call-ID: CXC-59-6942a370-a4a109cc-13c4-4bb0601c-2729e3f7-2afee549
  CSeq: 1 ACK
  Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hg4bk-287eb-4bb0601f-2729ee10-66a221b2
  Max-Forwards: 70
  P-Asserted-Identity: <sip:Anonymous@sip.skype.com>
⊕ Contact: <sip:Anonymous@204.9.161.164:5060;transport=udp>
  Content-Length: 0
```

# Outbound Call from IP-PBX (1/7)



AddPac IP-PBX Series



# Outbound Call from IP-PBX (2/7)

## (1) INVITE

```
⊕ Request-Line: INVITE sip:827079510919@sip.skype.com SIP/2.0
⊖ Message Header
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk964b554ca484
  ⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
  ⊕ To: <sip:827079510919@sip.skype.com>
  Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
  CSeq: 84 INVITE
  Supported: timer, 100rel
  Min-SE: 1800
  Date: Mon, 29 Mar 2010 17:36:38 GMT
  Session-Expires: 1800
  User-Agent: AddPac IP-PBX
  ⊕ Contact: <sip:99051000003457@60.196.6.77>
  Accept: application/sdp
  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, UPDATE, PRACK, REFER, NOTIFY, INFO
  Content-Type: application/sdp
  Content-Length: 449
  Max-Forwards: 69
⊖ Message body
  ⊖ Session Description Protocol
    Session Description Protocol version (v): 0
    ⊕ Owner/Creator, Session Id (o): 99051000003457 1269884196 1269884196 IN IP4 172.17.101.240
    Session Name (s): AddPac Gateway SDP
    ⊕ Connection Information (c): IN IP4 172.17.101.240
    ⊕ Time Description, active time (t): 1269884196 0
    ⊕ Media Description, name and address (m): audio 23394 RTP/SAVP 18 101
    ⊕ Media Attribute (a):ptime:20
    ⊕ Media Attribute (a):crypto:1 AES_CM_128_HMAC_SHA1_80 inline:wzF4/tpRiwLdXEzcioXzodD00ffSwJXMzmE7wrAX
    ⊕ Media Attribute (a):rtpmap:18 G729/8000
    ⊕ Media Attribute (a):rtpmap:101 telephone-event/8000
    ⊕ Media Attribute (a):fmtp:101 0-15
    ⊕ Media Description, name and address (m): audio 23394 RTP/AVP 18 101
    ⊕ Media Attribute (a):ptime:20
    ⊕ Media Attribute (a):rtpmap:18 G729/8000
    ⊕ Media Attribute (a):rtpmap:101 telephone-event/8000
    ⊕ Media Attribute (a):fmtp:101 0-15
```



# Outbound Call from IP-PBX (3/7)

## (2) 407 Proxy Authentication

```
⊕ Status-Line: SIP/2.0 407 Proxy Authentication Required
⊖ Message Header
⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
⊕ To: <sip:827079510919@sip.skype.com>;tag=a4a109cc-13c4-4bb0669a-27433f01-4d79c76
  Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
  CSeq: 84 INVITE
  Proxy-Authenticate: Digest realm="sip.skype.com", nonce="4bb066b80001781bc891b9609d299d2022c6c16fb79e8c19", algorithm=MD5
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hG4bK964b554ca484
  Content-Length: 0
```

# Outbound Call from IP-PBX (4/7)

## (3) ACK

```
⊕ Request-Line: ACK sip:827079510919@sip.skype.com SIP/2.0
⊖ Message Header
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk964b554ca484
  ⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
  ⊕ To: <sip:827079510919@sip.skype.com>;tag=a4a109cc-13c4-4bb0669a-27433f01-4d79c76
    Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
    CSeq: 84 ACK
    Content-Length: 0
    Max-Forwards: 70
```

# Outbound Call from IP-PBX (5/7)

## (4) INVITE

```
⊕ Request-Line: INVITE sip:827079510919@sip.skype.com;transport=udp;maddr=204.9.161.164 SIP/2.0
⊖ Message Header
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk964b554ca485
  ⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
  ⊕ To: <sip:827079510919@sip.skype.com>
  Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
  CSeq: 85 INVITE
  Supported: replaces, timer, 100rel, early-session
  Min-SE: 1800
  Date: Mon, 29 Mar 2010 17:36:38 GMT
  Session-Expires: 1800
  User-Agent: AddPac SIP Gateway
  ⊕ Contact: <sip:99051000003457@60.196.6.77>
  Accept: application/sdp
  Proxy-Authorization: Digest username="99051000003457", realm="sip.skype.com", nonce="4bb066b80001781bc891b9609d299d2022c6c16fb79e8c19",
  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, UPDATE, PRACK, REFER, NOTIFY, INFO
  Content-Type: application/sdp
  Content-Length: 449
  Max-Forwards: 70
⊖ Message body
  ⊖ Session Description Protocol
    Session Description Protocol Version (v): 0
    ⊕ Owner/Creator, Session Id (o): 99051000003457 1269884196 1269884196 IN IP4 172.17.101.240
    Session Name (s): AddPac Gateway SDP
    ⊕ Connection Information (c): IN IP4 172.17.101.240
    ⊕ Time Description, active time (t): 1269884196 0
    ⊕ Media Description, name and address (m): audio 23394 RTP/SAVP 18 101
    ⊕ Media Attribute (a):ptime:20
    ⊕ Media Attribute (a): crypto:1 AES_CM_128_HMAC_SHA1_80 inline:wzF4/tpRiWldxEzcioXzodD0offSwJXMzme7wrAX
    ⊕ Media Attribute (a): rtpmap:18 G729/8000
    ⊕ Media Attribute (a): rtpmap:101 telephone-event/8000
    ⊕ Media Attribute (a): fmp:101 0-15
    ⊕ Media Description, name and address (m): audio 23394 RTP/AVP 18 101
    ⊕ Media Attribute (a):ptime:20
    ⊕ Media Attribute (a): rtpmap:18 G729/8000
    ⊕ Media Attribute (a): rtpmap:101 telephone-event/8000
    ⊕ Media Attribute (a): fmp:101 0-15
```

# Outbound Call from IP-PBX (6/7)

## (5) 180 Ringing

```
⊕ Status-Line: SIP/2.0 180 Ringing
⊖ Message Header
⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
⊕ To: <sip:827079510919@sip.skype.com>;tag=a4a109cc-13c4-4bb0669a-27433f01-4d79c76
  Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
  CSeq: 85 INVITE
  User-Agent: sipgw-1.0
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk964b554ca485
⊕ Contact: <sip:827079510919@sip.skype.com:5060;maddr=204.9.161.164;transport=udp>
  Content-Length: 0
```

# Outbound Call from IP-PBX (7/7)

## (6) 200 OK

```
⊕ Status-Line: SIP/2.0 200 OK
⊖ Message Header
⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
⊕ To: <sip:827079510919@sip.skype.com>;tag=a4a109cc-13c4-4bb0669a-27433f01-4d79c76
  Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
  CSeq: 85 INVITE
  Allow: INVITE,ACK,CANCEL,OPTIONS,BYE
  User-Agent: sipgw-1.0
  Via: SIP/2.0/UDP 60.196.6.77:5060;branch=z9hg4bk964b554ca485
⊕ Contact: <sip:827079510919@sip.skype.com:5060;maddr=204.9.161.164;transport=udp>
  Content-Type: application/sdp
  Content-Length: 220
⊖ Message body
⊖ Session Description Protocol
  Session Description Protocol Version (v): 0
⊕ Owner/Creator, Session Id (o): 99051000003457 1269884196 1269884196 IN IP4 204.9.161.164
  Session Name (s): skype call
⊕ Connection Information (c): IN IP4 204.9.161.164
⊕ Time Description, active time (t): 0 0
⊕ Media Description, name and address (m): audio 24068 RTP/AVP 18 101
⊕ Media Attribute (a): rtpmap:18 G729/8000
⊕ Media Attribute (a): rtpmap:101 telephone-event/8000
⊕ Media Attribute (a): fmp:18 annex=no
```

## (7) ACK

```
⊕ Request-Line: ACK sip:827079510919@sip.skype.com;transport=udp;maddr=204.9.161.164 SIP/2.0
⊖ Message Header
  Via: SIP/2.0/UDP 60.196.6.77;branch=z9hg4bk964b554ca485
⊕ From: <sip:99051000003457@sip.skype.com>;tag=964b554ca4
⊕ To: <sip:827079510919@sip.skype.com>;tag=a4a109cc-13c4-4bb0669a-27433f01-4d79c76
  Call-ID: 9666b04b-146c-556e-804c-0002a4ff4869@60.196.6.77
  CSeq: 85 ACK
  Content-Length: 0
  Max-Forwards: 70
```



# Thank you!

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