

How to Build a Simple Virtual Office PBX System Using TekSIP

This document explains how to build a simple virtual office PBX system using TekSIP. In this example following components are used:

- **VoIP Gateway.** VoIP gateway is used for PSTN interfacing. AudioCodes MP-114 with FXO ports (*Version ID 5.00A.042.001*).
- **Soft IP Phone.** Xten eyebeam (*Version 1.1*).
- **IP Phone.** Cisco 7912 Series [*Software Version 1.02.00(040406A)*]
- **TekSIP** (*Version 1.9*).

You can see how virtual PBX components organized on the office LAN in the diagram below:

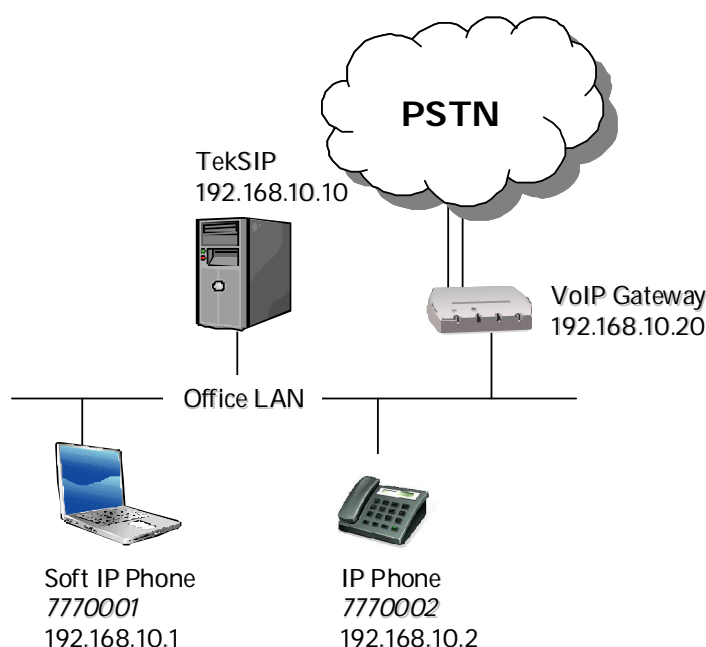


Figure - 1. Sample Topology

IP Phones (*Hard or soft*) on the office LAN can dial each other through TekSIP. TekSIP can be configured to divert a call, if destination is not on-line. You can also route calls to recipients not in TekSIP's registration database to a default SIP endpoint. You can route PSTN numbers to voice gateway which has PSTN connections. It's also possible that you can route incoming PSTN to local SIP endpoints.

TekSIP Configuration

TekSIP configuration is very simple. Select an IP address to be listened from detected IP address list. You do not need change default port number 5060. Although you should use a DNS registered domain name, in our simple virtual PBX system we can use IP address of SIP Proxy as SIP domain.

TekSIP can redirect calls which are directed to endpoints which are not in registered endpoints list. You can redirect such calls to the endpoint specified in "Default Route" ay Routing tab. As we'll use a SIP Gateway, default route type will be SIP UA. If your default gateway a SIP PROXY / Registrar which requires authentication, you can specify a username/password pair for authentication of outgoing calls from TekSIP. You can enable logging optionally. SIP endpoint and session authentication is enabled by default.

If you choose to use TekSIP with “Authentication Enabled”, you need to define SIP endpoints in “Endpoints” tab.

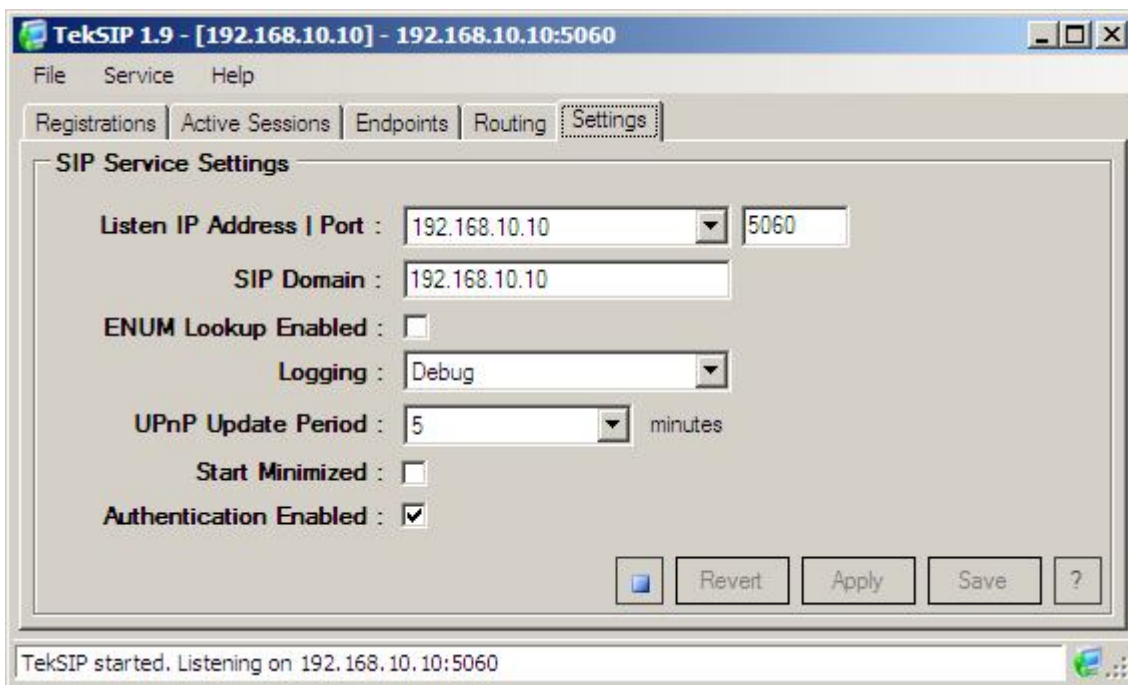


Figure - 2. TekSIP Configuration (*Settings*)

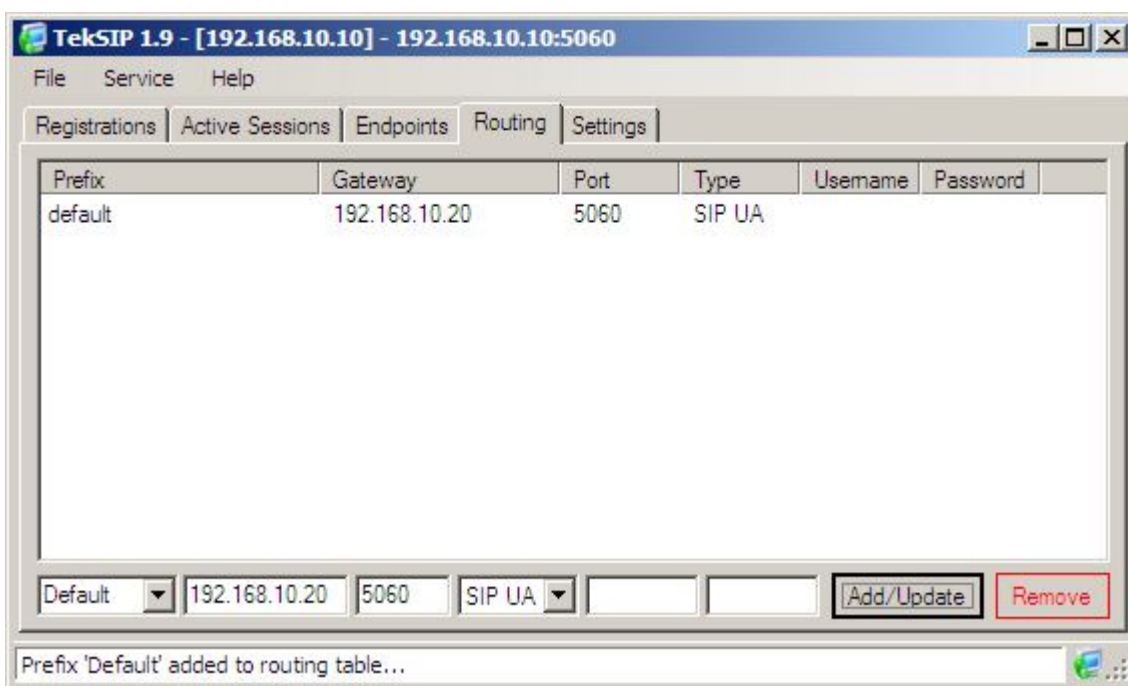


Figure - 3. TekSIP Configuration (*Routing*)

IP Phone Configuration (Cisco 7912)

Although there is remarkable number of IP phone alternatives in the market, we’ll use Cisco’s 7912 IP Phone device in our virtual office PBX. 7912 is one of the most common IP Phone devices.

Configure IP Phone as follows:

UseTftp	0
TftpURL	0.0.0.0
CfgInterval	3600
EncryptKey	•
EncryptKeyEx	•
Dhcp	0
StaticIP	192.168.10.2
StaticRoute	0.0.0.0
StaticNetMask	255.255.255.0
Domain	0
DNS1IP	0.0.0.0
DNS2IP	0.0.0.0
NIPIP	0.0.0.0
AltNIPIP	0.0.0.0
TimeZone	17
OpFlags	0x00000062
VLANSetting	0x0000002b
TOS	0x000060b8
NPrintf	0.0.0.0
TraceFlags	0x00000000

Figure - 4. Network Parameters

UID	7770002
PWD	•
Proxy	192.168.10.10
AltProxyTimeOut	0
UseLoginID	1
LoginID	7770002
SIPRegInterval	3600
MaxRedirect	5
SIPRegOn	1
NATIP	0.0.0.0
SIPPort	5060
MediaPort	18500
OutBoundProxy	192.168.10.10
MsgRetryLimits	0x00000000
NatServer	0
NatTimer	0x00000000
DialPlan	0
IPDialPlan	0

Figure - 5. SIP Parameters

Allow Call Waiting	0
Allow Call Transfer	0
Allow Blind Transfer	0
Allow Conference	0
Block Caller ID	0
Block Anonymous Calls	0
Do Not Disturb	0
Voice Mail Number	
Call Forward Number	
Call Fwd on Busy Number	
Display Name	7770002
Short Name	
Time Format	h:ia
Date Format	m-d-y
Forward to VMail Delay	20
GUI Show Mask	0xFFFFFFFF
GUI Set Mask	0xFFFFFFFF

Figure - 6. Call Preferences

Soft IP Phone Configuration

You can use any SIP soft phone in you virtual PBX system. You can see sample configuration for eyebeam:

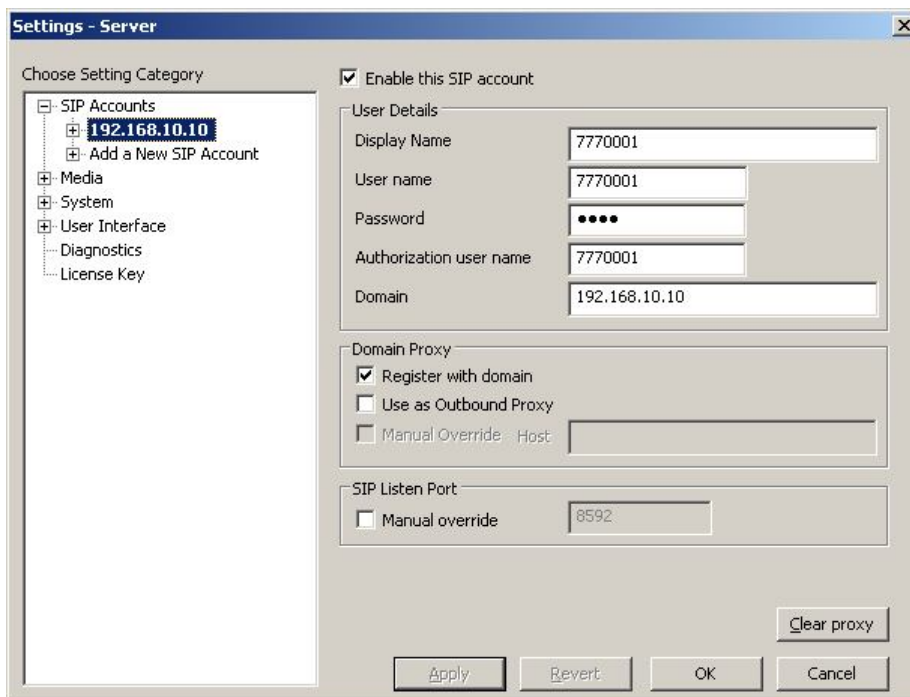


Figure - 7. eyeBeam Configuration

VoIP Gateway Configuration (AudioCodes MP-114)

An AudioCodes MP-114 is used in our Virtual PBX system to interface with PSTN. TekSIP is configured to forward calls to endpoints which are not in registration database to MP-114. Configure MP-114 to route calls to PSTN as follows:

Quick Setup	
IP Configuration	
IP Address	192.168.10.20
NAT IP Address	192.168.10.20
Subnet Mask	255.255.255.0
Default Gateway IP Address	
SIP Parameters	
Gateway Name	192.168.10.20
Working with Proxy	Yes
Proxy IP Address	192.168.10.10
Proxy Name	192.168.10.10
Enable Registration	Enable
Tables	
Coders Table	-->
Tel to IP Routing Table	-->
Endpoint Phone Number Table	-->

Figure - 8. MP-114 Quick Setup

Protocol Management / Hunt Group Settings

Hunt Group ID	Channel	Select Mode	Registration Mode
1	1	Dest Number + Cyclic Ascending	
2			
3			
4			
5			
6			
7			
8			
9			
10			
11			
12			

Protocol Management / Endpoint Settings / Authentication

Authentication		
Gateway Port	User Name	Password
Port 1 FXO	7771000	*****
Port 2 FXO	7771000	*****
Port 3 FXO	7771000	*****
Port 4 FXO	7771000	*****

Protocol Management / Endpoint Phone Numbers

Endpoint Phone Number Table				
	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1	1-4	7771000	1	0
2				
3				
4				

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