How to Build a Simple Virtual Office PBX System Using TekSIP and TekIVR

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

- **VoIP Gateway.** VoIP gateway is used for PSTN interfacing. AudioCodes MP-118 with FXO ports (*Version ID 6.60A.228.011*).
- Soft IP Phone. Xten eyebeam (Version 1.1).
- **TekSIP** (*Version 3.4.7*).
- **TekIVR** (Version 2.3.4).

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 when they register themselves to TekSIP. You can configure an alternative extension when the extension is not available to answer the call (*Off-line, busy, etc.*) while adding the extension. You can also route calls to recipients not in TekSIP's registration database to a SIP route based on called number. 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. Routing definitions can be made in Routing tab of TekSIP Manager.

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 will be used. 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. As we'll use a SIP Gateway, default route type will be SIP UA. You can enable logging optionally. SIP endpoint and session authentication is enabled by default.

🤕 TekSIP 3.4 - [192.168.1.4] - 192.168.1.4:5060 – 🗖 🗙
<u>File S</u> ervice <u>H</u> elp
Registrations Active Sessions Endpoints Routing Application Log Recordings
Service Parameters Accounting Authentication Services Counters
Service Parameters
Listen IP Address Port Transport : 192.168.1.4 💌 5060 UDP&TCF 💌
TLS Port Server Certificate : 5061 bke 💌 😭
SIP Domain : 192.168.1.4
Use External Address : 🔲 ?
UPnP Update Period : 3 vinutes
Enable STUN Server :
ENUM Lookup Enabled :
B2BUA for 3xx Responses :
Startup Mode Logging : Manual Debug
Save Registrations : 🔽
Revert Apply Save
TekSIP Service is started.

Figure - 2. TekSIP Configuration

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

🤕 Te	•kSIP 3.4 - [192.168.1.4] - 192.168.1.4:5060 – 🗖 🗙
<u>F</u> ile <u>S</u> ervice <u>H</u> elp	
Registrations Active Session	ns Endpoints Routing Application Log Recordings Settings
Endpoint	Redirect to Endpoint
🥑 10	None
ee 11	None
6 333	TOTO
Enter usemame	
TekSIP is started. Listening o	on 192.168.1.4:5060 [Accounting disabled] 🤤 📰

Figure - 2. TekSIP Endpoints

TekIVR Settings

You can use TekIVR as a call attendant for incoming calls from PSTN. If you define SIP extensions also in TekIVR and specify TekSIP as a SIP presence server, TekIVR will query presence status of defined extensions. This will enable TekIVR to check extension status prior to transfer a call. You can transfer the call to another extension or request a new extension from the caller. You should have prerecorded audio files or messages to be synthesized by TTS for your IVR scenario;

- A welcome message and request entry for an extension
- A waiting announcement to be played out while transferring the call
- A notification message if dialed extension busy
- A request for a new extension if dialed number is unavailable to receive the call

Basic settings for TekIVR is shown below. Please note that TekSIP account is also defined as a presence server. You should set Startup Mode = Auto. Logging should also be set to "None".

3	TekIVR 2.3 -	192.168.1.4:5070	- • ×
Elle Help Service Monitor 112 Estensions 22 V Service SIP Accounts Voice Mail Service Settings	laice Messages 🛛 📽 Settin Counters	94 🔣 Scenario Editor	
Listen IP Address Port	: 192.168.1.4	• 5070	
Presence Server	: teksip	•	
Use External Address	: 17		
UPnP Update Period	: 0 .	minutes	
Logging Startup Mode	: Debug 💌	Manual	
Transfer Method DTMF Transport	SIP REFER	RFC 2833	
Recognition Locale TTS Voice	English (United State *	Microsoft David Desktop	
Start Minimized	÷ 17		
HTTP Server Enable HTTP Server HTTP Server Port Login Password	: [* : [8282 : [*****		
			📓 Sare
This condition has been defined before.			Started 🍃

Figure - 3. TekIVR Settings

You need to define TekSIP as a SIP account.

8	TekIVR 2.3	- 192.168.1.4:5070	- = ×
Elle Help Service Monitor 517 E Service SIP Accounts SIP Account Settings	istensions 🖉 Voice Messages 🛛 Settin Voice Mail Counters	ge 丞 Scenario Editor	
SIP Accounts teksip	Gateway Port Usename Paasword Domain Register Timeout Ernabled Transport	192,168,1,4 5060 999 •••• 192,168,1,4 Yes 3600 Yes UDP	
Enter DF acceunt name	Gateway Select / set SIP Server for this accou	rts. Enter an IP address or a FQDN.	jai Save
This condition has been de	fined before.		Started

Figure - 4. TekSIP account definition in TekIVR

There must be an extension for TekIVR in TekSIP. Extension # 999 is defined for TekIVR in TekSIP. You should also define extensions in TekIVR.

8		Tek/VR 2.3 - 15	2.168.1.4:5070		- 🗆 🗙
Eile Help	19 Edensions & Voice	e Messages 📝 Settings	🔏 Scenario Editor	1	
Extension	Malbox Number	Send to E-mail	Enabled		
€ 10 € 11	20 21	yasin@kaplan.net hanfe.duma@kapla	Yes Yes		
Etter entermonen (B	Enteremal	address		Yes 💌 🛄 Add/U	odate X.Renove
his condition has t	been defined before.				Started 🦪

Figure - 5. Extensions in TekIVR

TekIVR Scenario

You need to define prompts to be used in your IVR scenario prior to create your IVR scenario. You can either use TTS or prerecorded audio files (*16 bit, 8 KHz, mono wave*).

TUTER	000 1003		
D 1 2 3 4 5 6 7 8	Turina Default Eraty Offine Ordine Bury Thanks GetExtension LeaveMessage Transferring	Welcome: This is a test prompt. C:\Program Ries (x85):Tek(VR:\Prompts':Kaplan Extension: a online C:\Program Ries (x85):Tek(VR:\Prompts':Kaplan C:\Program Ries (x85):Tek(VR:\Prompts':Kaplan C:\Program Ries (x85):Tek(VR:\Prompts':Kaplan C:\Program Ries (x85):Tek(VR:\Prompts':Kaplan C:\Program Ries (x85):Tek(VR:\Prompts':Kaplan	

Figure - 6. Prompts in TekIVR

You can create an IVR scenario using TekIVR graphical scenario editor. TekIVR initially has a default IVR scenairo with a Play and Transfer action defined in it. Select Delete \bowtie tool on tool bar

and delete connection between Play action and Transfer action by clicking on the connection. Please also delete Transfer action by clicking on it.

Click Select Tool on the tool bar. Select Play action in the scenario. Set properties of the Play action.

le Help Service Monitor 1} Extension	is 🖉 Voice M	lessages [1 Setting	a 📓 So	enam Edi	or				
hompts Scenarios	dPlay (Play Ex	ej Eval d	Wan E	#Exec Execute	Q-XFer Transfer	d'Rec Record	# Synt Synthesise	5 trat Status	¥ Edt Ext	* × 0
	471ay									
Called Nun Vaid Days Start Time 00:00 End Time 23:59 Maxemum 110	Digt Time Digt Long Prompt Count	out zh				2 2 1 Entry 1				

Figure - 7. First Play action in the scenario

Set Prompt parameter to your welcome message. Click Add Tool on the tool bar and drag and drop following actions to scenario screen; Status, Transfer, Play, Play.

		Te	kIVR 23	- 192.16	B.1.4:507	10				- ×
ile Help Service Montor [1] Extensions Prompts Scenarios) & Voic	e Messages	3	ings 🔝 S	icenam Ed	tor				
X 🖒 🔬 👷	d Play	+; Eval	di Wesh	#Exec	Q-XFer	d'Rec	d Synt	5 101	X DOL	10 14
Scenario Name	Play	Evaluate	Wat	Execute	Transfer	Record	Synthesise	Status	Ext	XI
Called Nue Vald Days Start Time 00:00 End Time 23:59 Maximum 1 10				d DXFer	Ray					
Called Number Specify called number for this scenario										
and the second	e de la conserv	6 2107 30	0							ALC: STORE OF

Figure - 8. Adding other actions

Click Link Tool on the tool bar. Select Play action in the scenario. Drag and drop to Stat action. This will link first play action to Status action.

Vaice Messages I Sr ay e2buil <u>a Wain</u> ay Evaluate Wat	etings 🕼 Scenato E #Exec (9-XFer Execute Transfer	dfor	4 Synt Synthesise	U Stat Status	Est	<u>* *</u> × Z
chey	-2 Play					
4 Stat	@May					
ateration	Terretorial	%receiv	ed_digita%			
nine Action ffline Action luey Action		-1 -1 -1				
resion fy an extension to query to	s presence status; set %	eceived_dig	ts% to transfe	r to the us	er daled num	ber.
	stension Vilne Action Milne Action Iusy Action Iusy Action Fision Ry an extension to query to	CARE AND	Affrestion Strective Value Action -1 Jusy Action -1 Iusy Action -1 resion for query its presence status; set "treceived_dig	Atension <u>Vreceived_digita%</u> Atension -1 Naine Action -1 luey Action -1 resion -1 resion to query to presence statue; set %received_digita% to transfe	Atension School of the set Sch	Atension Scher

Figure - 9. Linking first Play action to Status action.

Click link tool again, click Stat action and drag and drop to Xfer action. TekIVR will transfer the call if dialed extension is available to receive the call after this procedure. You should also specify a prompt in Transfer action properties.

Remice Montor 150 Extensions Prompts Scenarios	s 🚜 Vaice Mer	ssages 🚀 Sett	ings 👔 Scenario Ed	llor				
Scenario Name default	4Play s2 Play Eva	Eval d Wait	Evecute Transfer	& Rec Record	4 Synt Synthesise	4 Status	Est	
		6/ SCH 1						
		4 30H	d Hay O XFer					
Called Nur	Extension	* 30F	d May Q-XFer	%ласазім	ed_digits%			
Called Nun Valid Days	Estension Online Acts	an	@Hey OXFer	%receiv	ed_digita%			
Called Nun Valid Days Start Time 00:00 End Time 23:59 Maximum # 10	Estension Online Acto Offline Action Busy Action	en n	@ Play	2/recsiv -1 -1 -1	ed_digita%			

Figure - 9. Linking Status action to Transfer action

Click link tool again, click Stat action and drag and drop to second Play action twice. This will instruct TekIVR to play a notification message to caller party announcing that called party is unavailable to receive the call.

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le Help Service Montor 🕀 Extensio	rokiVK Z. ns 🖉 Voice Messages 📽 Sett	a = 192,106,1,400,	170 tor]				
Scenario Name	C 4Play 12 Eval 4 Wat Play Evaluate Wat	Albeet OrkFer Execute Transfer	& Rec Record	4 Synt Synthesise Synthes	V State Status ise audio b	Ed ased on use	N P
	4 Stat						
		@Play					
Called Nun Vaid Days Start Time 00:00 End Time 23:59 Maximum 1:10	Digit Timeout Digit Length Prompt Count	d Play	0 7 4 Busy 1				

Figure - 9. Linking Busy and Offline cases of Stat action to second Play action

Link second Play action to third Play action to request a new extension from the caller.

	TekIVR 2	13 - 192,168,1,4:5070			3	
ile Help Service Montor 150 Extensio Prompts Scenatos • • • •	ns & Vace Messages # Se	ttings 🕼 Scenario Editor #Exec QXFer CRec	4 Spec	4 Stat	• Bet	
Scenario Name	Play Evaluate Wat	Execute Transfer Recor	d Synthesise	Status	Ext	XI
	w son	@Play SXFer				
Called Nun	Digit Timeout	0				
Valid Days Start Time 00-00	Digt Length	7	2			
End Time 23:59 Maximum # 10	Count	1	5			
Called Number	Count	t a value between 1-30				
scenario						

Figure - 9. Linking second Play action to third Play action to request a new extension

Finally link third Play action to Stat to check presence status of the new dialed extension.

ns 🖉 Voice Messages 🗐 Setting	ga 👔 Scenarto Edit	or]			
4 Play ageval a Visit Play Evaluate Wat	Affeet Oxfee Execute Transfer	Record Sy	4 Synt 4 S	tut Ext	<u>*</u> * × 2
U Star	d Play				
Eggl: Tempour Digit Length Prompt Count	Lances	2 2 6 GetExtern 1	sion		3
	ns & Voice Messages & Settin	ns & Vaice Messages & Settings II Scenario Edit	Reverse and the second of the	Reverse and the second synthesise of the second second synthesise of the second sec	A Voice Messages Settings Scenario Editor

Figure - 10. Linking Third Play action back to Status action

Soft IP Phone Configuration

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

	Settings		×
Choose Setting Category SIP Accounts G· 192.168.1.4 G· Add a New SIP Account · Media · System · User Interface · Diagnostics · License Key	 Enable this SIP account User Details Display Name User name Password Authorization user name Domain 	10 10 •••• 10 192.168.1.4	
	Domain Proxy Register with domain Use as Outbound Proxy Manual Override Host		
	SIP Listen Port	8322 Clear pro	ху

Figure - 11. eyeBeam Configuration

VoIP Gateway Configuration (AudioCodes MP-118 FX_FXS)

An AudioCodes MP-118 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-118.

2 TekSIP	3.4 - [192.168.1.4] - 192	.168.1.4:5060 – 🗆 🗙
<u>F</u> ile <u>S</u> ervice <u>H</u> elp		
Registrations Active Sessions Er	ndpoints Routing Application I	Log Recordings Settings
Prefix	Gateway	192.168.1.8
default	Port	5060
	Туре	SIP UA
	Usemame	
	Password	
	Domain	192.168.1.8
	Register	No
	Registration Timeout	3600
	Remove Prefix	No
	Enabled	Yes
	Transport	UDP
	Gateway Select / set gateway for this rou	te. Enter an IP address or a FQDN.
Prefix : Enter a phone number p	refix or select default	▼ Add Route Celete Route
TekSIP is started. Listening on 192	.168.1.4:5060 [Accounting dis	abled] - [External IP Address : 78.180 🥑 🚲

Figure - 12. TekSIP default route

Configure MP-118 FXO_FXS to route calls to PSTN as follows. You must submit configuration changes, burn to flash and re-start the gateway after the configuration.

▼		
Use Default Proxy	No	v
Proxy Name	192.168.1.8	
Redundancy Mode	Parking	~
Proxy IP List Refresh Time	60	
Enable Fallback to Routing Table	Disable	~
Prefer Routing Table	No	~
Always Use Proxy	Disable	~
Enable Registration	Disable	~
Gateway Name	192.168.1.8	
Gateway Registration Name	192.168.1.8	
Subscription Mode	Per Endpoint	~
User Name	20	
Password	ż	
Cnonce	Default_Cnonce	
Registration Mode	Per Endpoint	¥

Figure - 13. MP-118 FXS_FXO, SIP Definitions / Proxy & Registration

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	Hunt Group ID	Channel Select Mode	Registration Mode
1	1	Cyclic Ascending	Don't Register 🗸
2		×	¥
3		v	v
4		×	¥

Figure - 14. GW and IP to IP / Hunt Group / Hunt Group Settings

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	5-8		1	0
2				
3				
4				
5				
6				

Figure - 15. GW and IP to IP / Hunt Group / Endpoint Phone Number

	Src. Hunt Group ID	Dest. Phone Prefix	Source Phone Prefix	- >	Dest. IP Address	Port	Transport Type	Dest. IP Group ID	IP Profile ID	Cost Group ID
1	*	*	*		192.168.1.4	5060	UDP 🗸	-1	0	None 🗸
2							Not Configured 🗸	-1		None 🗸
3							Not Configured 🖌	-1		None 🗸

Figure - 16. Routing / Tel to IP Routing

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	IP Profile ID
1	*	*	192.168.1.4		1	0
2						
3						

Figure - 17. Routing / IP to Hunt Group Routing

Gateway Port	Destination Phone Number	Auto Dial Status	Hotline Dial Tone Duration [sec]
Port 1 FXS		Enable 🗸	0
Port 2 FXS		Enable 🗸	0
Port 3 FXS		Enable 🗸	0
Port 4 FXS		Enable 🗸	0
Port 5 FXO	999	Enable 🗸	0
Port 6 FXO	999	Enable 🗸	0

Figure - 18. Analog Gateway / Automatic Dialling

▼		
Dialing Mode	One Stage	*
Waiting for Dial Tone	No	¥
Time to Wait before Dialing [msec]	1000	
Ring Detection Timeout [sec]	8	
Reorder Tone Duration [sec]	255	
Answer Supervision	No	~
Rings before Detecting Caller ID	1	v
Send Metering Message to IP	No	~
Disconnect Call on Busy Tone Detection (CAS)	Enable	*
Disconnect On Dial Tone	Disable	~
Guard Time Between Calls	1	
FXO Double Answer	Disable	~
FXO AutoDial Play BusyTone	Disable	~
FXO Ring Timeout [100 msec]	0	

Figure - 19. Analog Gateway / FXO settings

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