

# Design and implementation of analog telephone on IPPBX network interconnection

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**Abstract.** The use of PABX between buildings existed in State Polytechnic of Malang consists of 3 types, which are produced by the same manufacturer. However, the three types of PABX have different characteristics and specifications from one another. Therefore, its merging is done by numbering method which must use prefix to be able to contact users in other PABX. Every PABX is always provided with an interface that aims to combine with different systems, so that adjustments can be made. To overcome this problem, by software-based central installation, namely IPPBX, so that if the change (migration process) occurs, a device called the internet telephone gateway can be installed. This device with one IP address can connect more than one analogue telephone number. So far the product has 32 ports. This research uses ITG with 16 ports due to the connection characteristics of the same connector as the conventional PABX. This condition will be easily implemented and does not change the existing network system in State Polytechnic of Malang. The results of the design based on the number of connected services required a number of 32 devices ITG 16 ports plus 4 ports connection to the public services telephone network (PSTN).

## 1. Introduction

The PABX combined between buildings existed in State Polytechnic of Malang, are produced by the same manufacturer. The first modular design of the PABX Super Hybrid System and 8 lines to as many as 448 extensions and 192 CO lines by adding expansion cards (512 ports maximally). The system can be installed as one, two or three shelf configurations, providing economical expansion from 192 ports (1 cabinet) to 384 ports (2 cabinets). The second and third PABXs use the Pure IPPBX type, but differ in capacity as shown in table 1 and table 2 [1].

**Table 1.** Type and maximum number of slots.

Slot Type		Maksimum number	
		KX TDE100	KX-TDE200
IPCMPR Card Slot		1	1
Free Slot		6	11
Virtual Slot	Virtual Trunk Slot	4	4
	Virtual Extantion Slot	4	4



**Table 2.** Maximum trunks and extensions.

Type	KX-TDE100	KXTDE200
Total Number of Trunks	128	128
Trunk (Physical Trunk Card)	120	128
Trunk (Virtual Trunk Card)	32	32
Total Number of Extensions	256	256
Extension (Physical Extension Card)	160	256
Extension (Virtual Extension Card)	128	128
IP-PT and IP Softphone	64	64
SIP Extension	128	128

Therefore, its merging is done by numbering method which must use prefix to be able to contact users in other PABX. Every PABX is always provided with an interface that aims to combine with different systems, so that adjustments can be made.

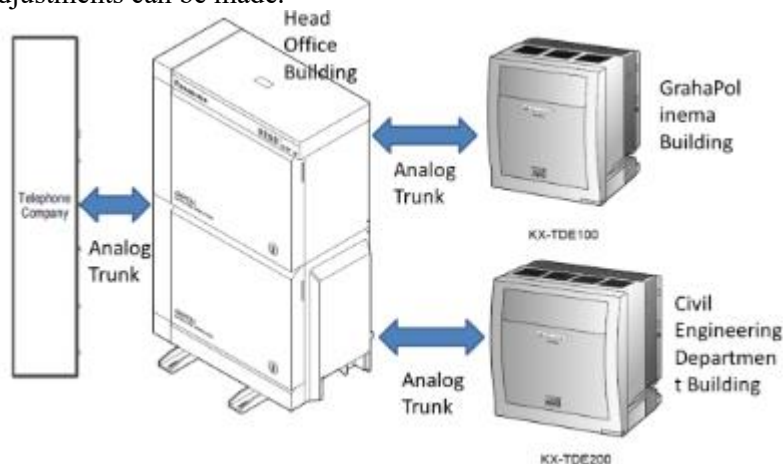
**Figure 1.** Existing PABX interconnection.

Figure 1 above shows the connection between the PABXs that are existed until now. The PABX installed in the Central Office Building consists of 16 ESLC cards, 1 LCOT card, 1 DHLC card and 1 DISA card. With this capacity, it can serve 256 analogue telephone users, 8 digital telephones plus 8 analogue telephones and 8 ports that can be connected to public telephone exchanges. The PABX installed in the Graha Polinema Building consists of 4 SLC16 cards, 1 LCOT8 card, 1 DLC8 card. With this capacity, it can serve 64 analogue telephone users, 8 digital telephones and 8 ports that can be connected to other telephone exchanges.

The PABX installed at the Civil Engineering Building consists of 7 SLC16 cards, 1 LCOT16 card, 1 DLC16 card. With this capacity, it can serve 112 analogue telephone users, 16 digital telephones and 16 ports that can be connected to other telephone exchanges. The problem that occurs is the limited connection between users in each building with different PABX services. Telephone users at the Graha Polinema building and civil engineering department building can only have conversations with telephone users with the central office of PABX service.

To solve this problem, the PABX is replaced by the internet protocol private branch exchange (IPPBX) [2], so that all processes can be done automatically, by operating application software via the WEB Graphic User Interface [3].

Then without changing the existing cable network and telephone equipment, an IPPBX server can centrally be connected through the intranet computer network.

The application uses an internet telephone gateway, each of which consists of 16 ports instead of ESLC or SLC16 cards as much as the capacity has been installed with the basic network architecture as shown below [4].



**Figure 2.** Network model offered.

## 2. Method

The design and implementation in this research includes:

- Selecting Internet Protocol Private Branch Exchange (IPPBX) devices that can register up to 800 users or more.
- Designing and selecting the internet telephone gateway device that matches the previously replaced card.
- Designing and implementing telephone networks using a computer network that is already installed.

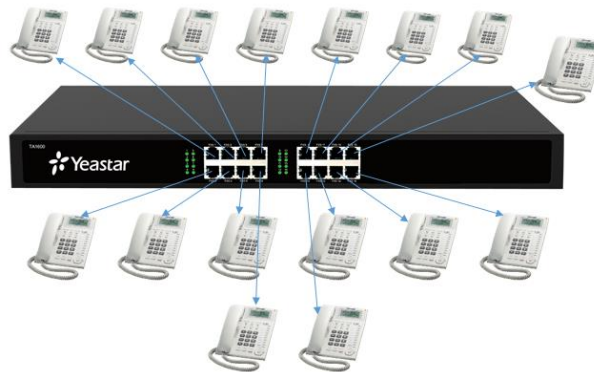
### 2.1. Required device specifications

The IPPBX comes with an asterisk-based system, the IPPBX software, offering not only full PBX functionality, but also a new feature that enables new stability for your unified communication systems. It can seamlessly integrate VoIP trunks and your existing PSTN lines with 4 port analog connections and 2 Ethernet ports. They are developed with a wide selection of codecs and signaling protocols, including G711 (alaw/ulaw), G722, OPUS, AMR-NB/WB, SILK, G723.1 G726, G729, GSM, ADPCM, iLBC, H263, H263P, H264, VP8 [5]. Taking full advantages of open source platform, the IPPBX Series appliances support industry standard SIP trunks, IAX2 trunks, analog PSTN trunks, and analog station trunks. Moreover, with IPPBX system software, the fully featured IPPBX is an ideal solution for the office. Detailed device specifications are shown in table 3.

**Table 3.** Specification of IPPBX.

Item	Description
System Capacity	Up to 800 extension register 100 Concurrent calls with G.729 codec 300 concurrent calls with G.711 codec
Max Network Interface	2x10/100M LAN port
Max FXS/FXO Interface	8
USB Port	1xUSB 2.0 for external storage or disaster recovery system
External Storage	1xSD slot, support up to 128G
Telephony Interface	FXS/FXO interface, Optional
RAM	DDR3 1GB
Storage	16GB Onboard Flash
Power Consumption	12V/1.33A 16W maximum
Followed Function	Enable in any combination of FXO and FXS modules

To integrate with other IPPBX, configuring the VoIP settings in Telephone Adapter (TA) FXS Gateway is needed to set up VoIP trunk (SIP and IAX).



**Figure 3.** Telephone Adapter with 16 port FXS.

## 2.2. VoIP server settings

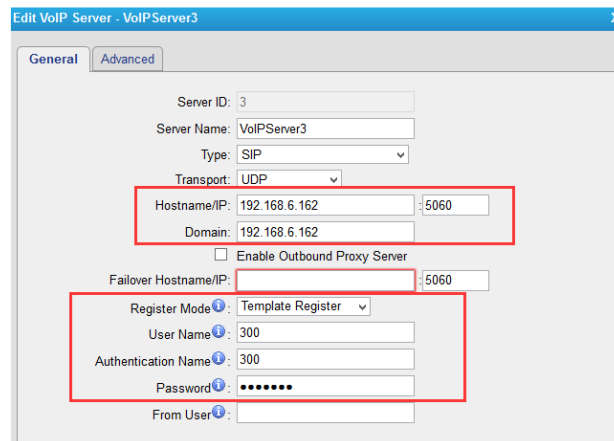
There are some configurable VoIP (SIP/IAX) Server templates on this page. The number of VoIP Server templates is the half of FXS ports on TA FXS Gateway. The VoIP server settings help the FXS ports to register to the VoIP server. After being configured, the templates can be chosen on FXS port setting page. There are 3 register modes for VoIP server. Users could select one mode for the VoIP server and apply it to FXS ports.

- Service Provider - IP Based VoIP Provider, do not generally require the TA gateway to register with the provider. Only IP address or domain is needed to configure on TA gateway.
- Port Register – the VoIP server requires TA gateway to register with the provider
- using an authentication ID and password. If you choose the VoIP server, you need to fill in User Name, Authentication Name and Password to register the FXS port.

 The screenshot shows a web-based configuration interface titled "Edit FXS Port - 1". It has two tabs: "General" (selected) and "Other Settings". Under the "General" tab, there are two input fields: "Caller ID Name" with the value "301" and "Caller ID Number" with the value "301". Below these is a section titled "VoIP Server Template". This section contains a dropdown menu for "VoIP Server" set to "pbx(2)". Below the dropdown are four input fields: "User Name" (301), "Authentication Name" (301), "Password" (masked with dots), and "From User" (empty). A red rectangular box highlights the "VoIP Server" dropdown and the "User Name", "Authentication Name", and "Password" fields.

**Figure 4.** Port register.

- Template Register – register to your VoIP server and apply the template to FXS ports. The FXS ports will register to the server with the same account.

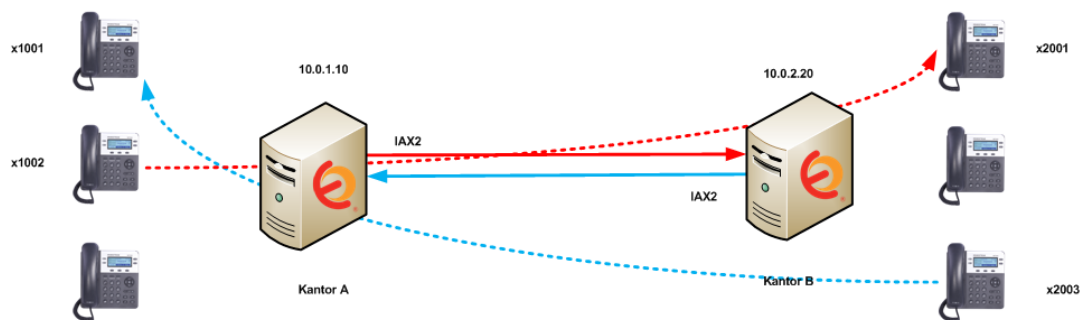


**Figure 5.** VoIP server.

**2.2.1. Architecture of plan.** This setting is to connect two or more servers that use Asterisk IP PBX (using Elastix version 4.0), so that dialling between extensions on both servers can be done easily, namely by simply pressing the extension number directly [6]. This method can still be used for 2 to 7 servers. The number of trunks that must be set up to produce a full mesh trunk network can be obtained by the formula:

$$n \times (n - 1)$$

Therefore, if we want to connect 7 IP PBX at once, it will take as much as  $7 \times (7-1) = 42$  trunks.



**Figure 6.** Intercommunication between two IPPBXs.

What needs to be prepared is as follows:

- If the two sites are on two different networks / places / cities, it is recommended that connections between the sites use VPN (PPTP, L2TP, Open VPN, VPLS, MPLS, EoIP etc.). And if you really have to use a public IP address, put the IP PBX server under the router/ firewall and just do port forward (not 1: 1 Network Address Translated). Trunk using IAX2, port forwarding is easy because only one port is needed for signalling and media streams, the TCP / UDP 4569 port.
- Extension number planning is very important because it is to facilitate the routing of calls. If you have two IP PBXs that you want to interconnect, use different extension heads, for example server A uses 4 digits with head 1 (1001, 1002, 1003, etc.), then server B uses 4 extensions with head 2 (2001, 2002, 2003, etc.) As an additional suggestion, or if the site reaches more than 7 and will still use interconnection this way, add extension digits, for example to 5 or 6 digits. Later it will be very helpful if we want to use additional prefixes (for example to trunk to PSTN / ITSP), or the number of sites reaches more than 10 sites.

- The trunk that will be used is IAX2 if each IP PBX is Asterisk based. If not, you can use SIP. The reason for using IAX2 is because it only uses one port for signalling and media streams (SIP uses a different signalling port rather than the media stream), so that setting up a firewall is quite easy. Besides, IAX2 is indeed designed to overcome the NAT problem.

### 3. Results and discussion

The results applied two IPPBX interconnections. This system was divided into two groups of users. The first group was with 1XXX number of 216 extensions, and the second with 2XXX number of 301 extensions that could be registered. It was shown in Figure 5 above. The settings of each IPPBX are illustrated below.

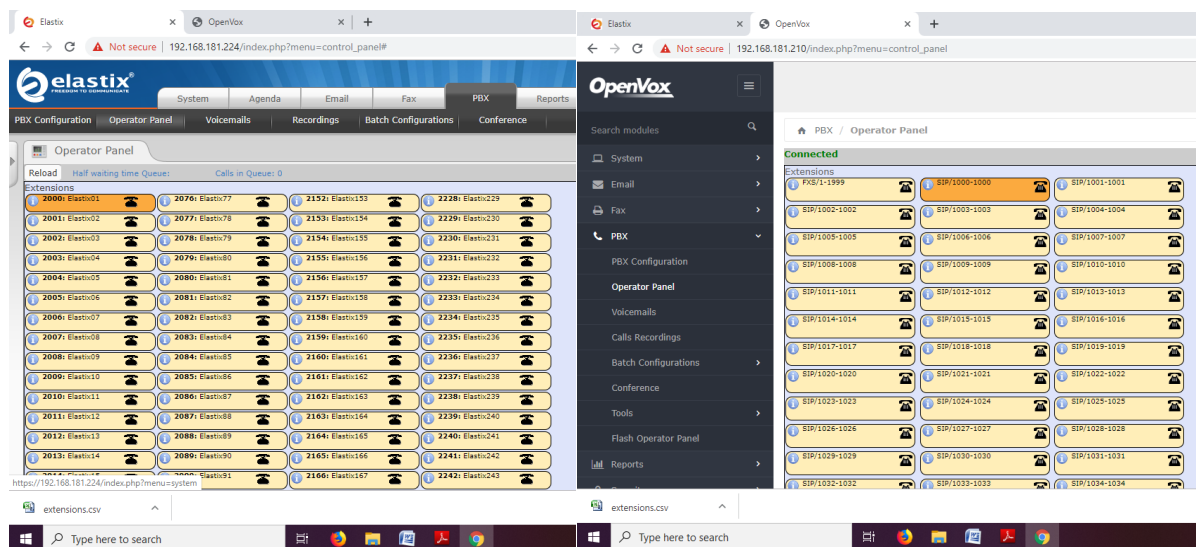


Figure 7. Shows the results of making extensions on the panel operator.

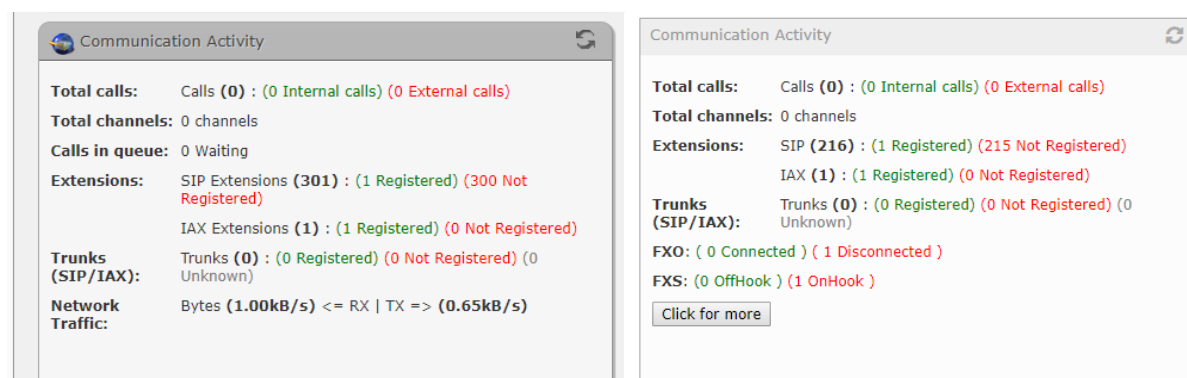
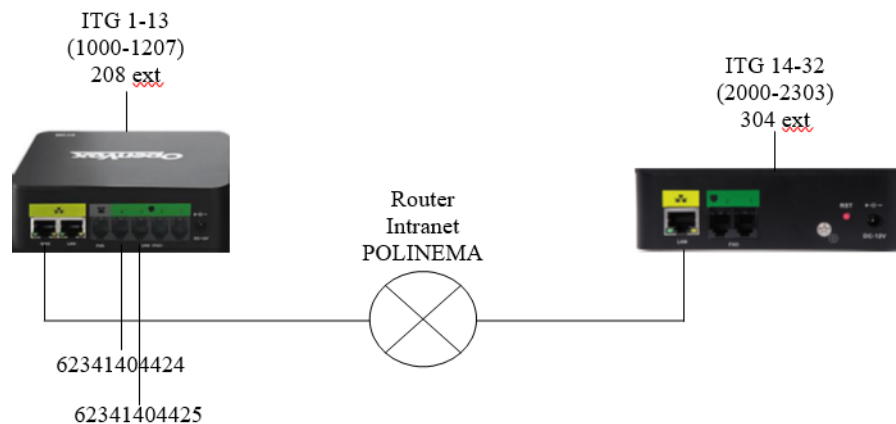


Figure 8. Shows communication activities.

The complete diagram of the results is described below.



**Figure 9.** Diagram of connection of full system design.

#### 4. Conclusion

- This research uses ITG with 16 ports due to the connection characteristics of the same connector as the conventional PABX. This condition will be easily implemented and does not change the existing network system in State Polytechnic of Malang. The results of the design based on the number of connected services required a number of 32 ITG devices 16 ports plus 4 ports connection to the Public Service Telephone Network (PSTN). However, only 2 ports are used.
- Some cables on the previous network could still be used.
- A mobile phone can be used as an extension.

#### References

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