VOICE OVER INTERNET PROTOCOL (VoIP): A VIEW AND A VISION

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Abstract

The aim of this study is to explain the fundamentals of VoIP, regarding the functions and components and how it can be deployed by various organizations as an effective way of communication. In essence this study will try to answer the following questions: What does it mean to an organization to deploy VoIP? What makes up a VoIP solution and how can they take advantage of it? .

Cisco router 9600 series and its peripheral devices such as VICs and Voice Network Modules in addition to the required router configuration software were used to verify the efficiency and reliability of VoIP over PSTN. The process used revealed that users will be able to send and receive calls as they would at same office or area, any where in the world, at no extra cost as call go via IP. This does not incur charges as call diversion does via PSTN, and the called party does not have to pay for the call. Once a general understanding of VoIP is achieved, organizations are better prepared to tackle the more complex issues that go into deploying a secure, low-cost, reliable and high-performance VoIP network.

Keywords:- VoIP, PSTN, VICs, FXS, FXO, PBX, E&M, IP Telephony.

INTRODUCTION

Voice over IP ('VoIP') is an important topic in information and communication technology circles, and increasingly within contact centers. From a world without VoIP just a few years ago, it is now becoming inevitable that VoIP will be deployed in everything from carriers' backbones to our offices and homes. Indeed, leading industry analysts project that within the next four years traditional call center telephony infrastructure suppliers like Avaya, Alcatel, Nortel Networks and Siemens will announce their intention to discontinue support within five years for system architectures based on Time Division Multiplexing (TDM)[1]. The move from traditional circuit-switched to packet switched network technology will have a significant business impact on the applications used to record customer, supplier and other interactions, and the associated benefits such applications can deliver. The term "Voice over Internet Protocol (VoIP)" is used to refer to the practice of conveying live voice communication (including fax) via packet-switching technology utilizing Internet Protocol and transmitted over public or private data networks [3]. This term thus includes the transmission of voice calls over the interconnected "public Internet" as well as the use of IP technology to convey voice calls over private leased point-to-point data networks [9]. VoIP technology has the technological advantage that it is an easy to deploy technology, that, when coupled with readily available Internet connections, allows easy and cost effective carriage of voice communication over the Internet. In addition, VoIP technologies make use of packet-switching technology, which is more efficient in its use of network capacity than circuit switched technology. The result is that with VoIP voice traffic is carried more efficiently over data networks, which generally cost users much less to use than circuit switched network, this study will enable smaller businesses to access application functionality previously afforded only by large organizations, yet also affording larger customers greater flexibility in their infrastructure with resulting cost savings. IP Telephony is also enabling the development of enterprise-wide contact management systems, which embrace every department that touches the customer, and every department that deals with the outside world. These external interactions with customers, suppliers and other third parties can include enormous amounts of valuable information and insight that, if made easily available to the right people in your business, may dramatically improve the service that you offer to your customers, and the overall business performance. Until now, however, the valuable insight contained in these telephone calls has been, at best, only captured and shared within the contact center. In this study I have showed the possibilities of deploying VoIP technology in Jordanian private and public sectors by covering the fundamentals and advance need to configure this technology in small business. This has been developed using Cisco router and other material as hardware equipments and Cisco router IOS Software commands for necessary configurations, this study as well contain a real practical example shows in reality the overall architecture of VoIP technology, and real implementations and configurations needed to deploy this technology, and shows it's interaction with public switching telephone network (PSTN), in addition to solutions to challenges of deploying this technology it real telecommunication systems for more effective and smoothly use.

VoIP IMPLEMENTATION CHALLENGES

Because IP does not by default provide any mechanism to ensure that data packets are delivered in sequential order, or provide any Quality of Service guarantees, implementations of VoIP face problems dealing with latency and possible data integrity problems[3].One of the central challenges for VoIP implementers is restructuring streams of received IP packets, which can come in any order and have packets missing, to ensure that the ensuing audio stream maintains a proper time consistency. Another important challenge is keeping packet latency down to acceptable levels, so that users do not experience significant lag time between when they speak and the signal is decoded on the other end of the connection.

Solutions to these problems include:

Certain hardware solutions can distinguish VoIP packets and provide priority queuing for this class of service.

Alternatively packets can be buffered but this can lead to an overall delay similar to that encountered on satellite circuits.

The network operator can also ensure that there is enough bandwidth end-to-end to guarantee low-latency low-loss traffic: this is easy to do in private networks, but much harder to do in the public Internet.

Jitter (delay variance) problems are mainly generated in lowband access (less than 256 kbit/s) because of serialization of big (1500 bytes) data packets. At these rates, fragmentation mechanisms for these big packets are needed (interleaving small voice packets) to reduce the delay. Over networks slower than 256 kbit/s it is almost impossible to ensure quality voice without a fragmentation mechanism.

HARDWARE CONFIGURATION

The two types of hardware of VoIP feature for Cisco 3600 series routers uses:

• Voice network modules
• Voice interface cards (VICs)

The configuration of these types of hardware should be taken (voice network module and VIC) before performing software configuration tasks. Voice network modules install in a slot in a Cisco 3600 series router, and convert telephone voice signals into a form that can be transmitted over an IP network. The one-slot voice network module provides one slot for a voice interface card. The two-slot voice network module provides two slots for voice interface cards. Voice interface cards install in slots in the voice network module, and provide connections to the telephone equipment or network. The VICs interfaces include:

Foreign Exchange Station (FXS), this interface connects directly to a standard telephone.

Foreign Exchange Office (FXO), this interface connects local call to a public switching telephone network, or to Private Branch Exchange (PBX).

Ear and Mouth (E&M), signaling technique for two-wire and four-wire telephone and trunk interfaces. The E&M VIC connects remote calls from an IP network to a PBX for local distribution.

To connect VICs to network, I used a standard Rj-11 modular telephone cable to connect FXS VIC ports to telephone or fax then I used Rj-11 modular telephone cable to connect FXO VIC ports to the PSTN or PBX, through a telephone wall outlet, and the E&M VIC uses RJ-48S connector and cable.

IOS SOFTWARE CONFIGURATION MODE

The main thing to be considered before entering to software configuration mode is the voice port numbering, to display the available ports numbering after finishing hardware configuration, is to use IOS show voice port command. To be able to use IOS configuration mode, first connects the console cable to the router then power up the router, after a few seconds, the EXEC prompt will be display. Type enables then types the password to enter to enable mode

Router> **enable**
Password:

As being familiar to router configuration, the configuration change can be made only by using enable mode. To enter to configuration mode
Router# **config terminal**
Router(config)#
If the router is not being configured before, the routing protocols should be configured as a follow

Router(config)# **ip routing**
Router(config)# **appletalk routing**
Router(config)# **ipx routing**

These configurations are general to all router configurations, and should be performed before proceeding in further configurations. In this time the Cisco 9600 router will be ready for VoIP necessary configuration.

FXS INTERFACES CONFIRGURATION

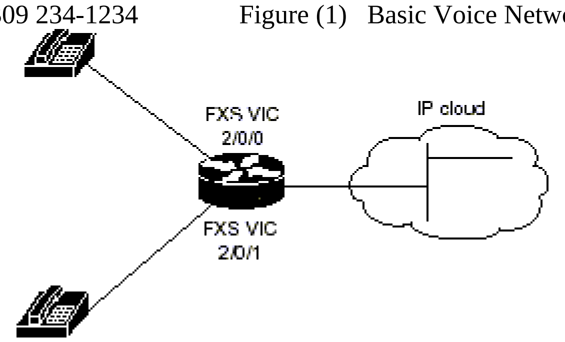
As I have mentioned before, the ports on FXS VICs connected directly to a standard telephone, fax machine, or similar device.
Before configuring FXS interfaces, and to make this section and other sections easy to understand , I have used the Philadelphia 1 as router hostname, and other routers will be naming sequentially Philadelphia 2, Philadelphia 3 … etc.

Table 1: Philadelphia 1 Router Telephone Numbers and Voice Ports

Telephone Number	Voice Port
309 234-1234	2/0/0
309 234-1456	2/0/1

309 234-1234

Figure (1) Basic Voice Network (Philadelphia 1)



309 234-1456

additional telephones and fax machine could be connected in this example, up 12 in total in case of using Cisco 3640 router, the remaining voice network module provide interfaces for IP connectivity to the LAN or WAN and for data traffic, and for more than 12 devices, we cam add more routers, or to use an E&M VIC and a local PBX, rather than connecting every telephone to its own FXS VIC. To

make routing a received voice call to the right destination, the router needs to know which telephone number belongs to each voice port. In other words, router needs to know the information in table 1.
To hold the information on the table 1, Cisco IOS software uses objects called dial peers. the information including voice port, telephone numbers, and other parameters for call are put together associating them all with the same dial peer.
Dial peers identified by numbers, but to avoid confusing these numbers with telephone numbers, they are usually referred to as a tags. Dial peers tags are integers that can range from 1 to $2^{31}-1$ (2147483647) Dial peers on the same router must have unique tags, but we can use reuse the tags on other routers.
Table 2 assigns each telephone number-voice port pair on the Philadelphia 1 router a dial peer tag. Within the allowed range, we can choose any dial peer tag or any system that is convenient. This type of dial peer is called a plain old telephone service (pots) dial peer or local dial peer means that the dial peers associates a physical voice port with a local telephone device.

Table 2: Philadelphia 1 Router Local Dial Peers

Telephone number	Voice Port	Dial Peer Tag
309 234-1234	2/0/0	211
309 234-1456	2/0/1	213

Now using the global configuration mode, I configure the router with the information in above table.
Philadelphia1 (config)# **dial-peer voice 211 pots**
Philadelphia1 (config-dial-peer)# **dest-pat +13092341234**
Philadelphia1 (config-dial-peer)# **port 2/0/0**
Philadelphia1 (config)# **dial-peer voice 213 pots**
Philadelphia1 (config-dial-peer)# **dest-pat +13092341456**
Philadelphia1 (config-dial-peer)# **port 2/0/1**
Philadelphia1 (config-dial-peer)# **exit**
Philadelphia1 (config)#

Cisco IOS software refers to a telephone numbers as a destination pattern, because it is the destination pattern for an incoming or outgoing call. A destination pattern always begins with (+) signs. It can also include asterisks (*) and pound signs (#) from the telephone keypad, and commas (,) and periods (.), which have special meaning. Parentheses (()), hyphens (-), slashes (/), and spaces ( ), which are often used to make telephone numbers easier for humans to read, are not allowed.
I supposed that Philadelphia has launched a new branch in south Jordan. Figure(2) shows the Philadelphia 2 network, and table 3 list phone numbers, voice pots, and dial peer tags for this branch.

Figure (2) Basic Voice Network (philadelphia2)

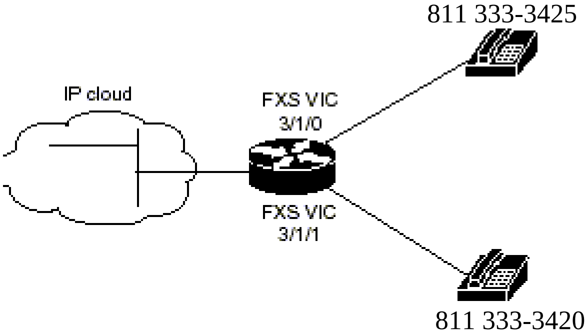


Table 3 Philadelphia 2 Router Local Dial Peers

Telephone Number	Destination Pattern	Voice Port	Dial Peer Tag
811 333-3425	+18113333425	3/1/0	801
811 333-3420	+18113333420	3/1/1	802

Philadelphia2 (config)# **dial-peer voice 801 pots**
Philadelphia2 (config-dial-peer)# **dest-pat +18113333425**
Philadelphia2 (config-dial-peer)# **port 3/1/0**
Philadelphia2 (config)# **dial-peer voice 802 pots**
Philadelphia2 (config-dial-peer)# **dest-pat +18113333420**
Philadelphia2 (config-dial-peer)# **port 3/1/1**
Philadelphia2 (config-dial-peer)# **exit**
Philadelphia2 (config)#

In office PBXs are usually configured so a user can dial a local call (within the same PBX) by dialing the extension only—for instance, the four-digit extension 3425 or the five-digit extension 43425 instead of the full telephone number, 18113333425. I provide the same shortcut on a voice over IP network by using the **number-expansion** (**num-exp**) command. This command tells the router to expand a particular sequence of dialed numbers into a complete telephone number (destination pattern). For instance, to expand 3425 into 8113333425, enter the following command:

Philadelphia2 (config)# **num-exp 3425 +18113333425**

To expand 4141 into 1 408 555-4141, enter the following command:

Philadelphia2 (config)# **num-exp 4141 +14085554141**
More generally, you can use the period (.) as a wild-card character representing a single digit. For instance, the command
Pheladelphia2 (config)# **num-exp.... +1408555....**
Expands any dialed sequence of four digits by prefixing +1408555 to it.
To use five-digit extensions beginning with the numeral 5 rather than four-digit extensions, you would enter the following command:
Philedlphia2 (config)# **num-exp 5.... +1408555....**
The corresponding commands for the East router would be (for four-digit extensions):
Philadelphia2 (config)# **num-exp .... +1919958....**
Or for five-digit extensions:
Philadelphia2 (config)# **num-exp 8.... +1919958....**
Now it is possible to place calls between telephones connected to same router using extension instead of using the full telephone number.
Naturally, the both branches would like to send voice traffic to each other over the same IP network they use for data traffic. A WAN port of some type on each router connects to the IP WAN. As we know the IP routers know how to locate IP addresses on the network, but they do not know how to

locate telephone numbers. to route an outgoing voice call over routers connection. The Philadelphia2 router has to associate a telephone number in the Philadelphia1 with the IP address of the Philadelphia1 router.
This is done by associating both pieces of information with a remote dial peer or VoIP dial peer on the philadelphia2 router as shown in table 4:

Table 4: philadelphia2 Router Remote Dial Peers

Remote Location	Telephone Number	Destination Pattern	IP Address	Dial Peer Tag
Philadelphia1	811 333-3425	+18113333425	192.168.16.1	601
Philadelphia1	811 333-3420	+18113333420	192.168.16.1	602

It is easier to use periods as a wildcard as in the following table:
Table 5: philadelphia2 Router Remote Dial Peers with Wild Cards

Remote Location	Telephone Number	Destination Pattern	IP Address	Dial Peer Tag
Philadelphia1	811 333-****	+1811333….	192.168.16.1	601

Now enter to the philadelphia2 global configuration mode and enter the following commands:

Philadelphia2 (config)# **dial-peer voice 601 voip**
Philadelphia2 (config-dial-peer)# **dest-pat +1811333....**
Philadelphia2 (config-dial-peer)# **session-target ipv4: 192.168.16.1**

In Cisco IOS software the remote network is known as session target. In this example this command followed by the IP address of the remote router. We can use the prefix dns followed by DNS name, for instance:
Philadelphia2 (config-dial-peer)# **session-target dns: voice.philadelphia1router.com**
The thing can be easier by configuring number expansion for philadelphia1 telephone numbers on the philadelphia2 router:

Philadelphia2 (config)# **num-exp 801.... +1811333....**

Now the user in philadelphia2 branch can dial a five-digit extension beginning with 8 to reach a telephone on the philadelphia1 area.
Now the philadelphia2 router is configured to send calls to the philadelphia1 router, this is shown in table 6:

Table 6 Philadelphia1 Router Remote Dial Peers with Wild Cards

Remote Location	Telephone Number	IP Address	Dial Peer Tag
Philadelphia2	309 234-****	192.168.20.1	701

Here it is good to configure RSVP on the WAN interface, and must configure each VOIP dial peer to request an RSVP session, to configure RSVP on WAN interface I used following commands:
Philadelphia1> **enable**
Password:
Philadelphia1# **configure terminal**
Philadelphia1 (config)# **interface serial 0/0**
Philadelphia1 (config-if)# **ip rsvp bandwidth**

Philadelphia2> **enable**
Password:
Philadelphia2# **configure terminal**
Philadelphia2 (config)# **interface serial 0/0**

Philadelphia2 (config-if)# **ip rsvp bandwidth**

Philadelphia1 (config-dial-peer)# **req-qos controlled-load**

Philadelphia2 (config-dial-peer)# **req-qos controlled-load**

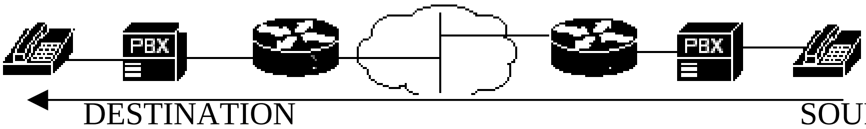
Otherwise no bandwidth is reserved for voice traffic.
If the bath between endpoints of a voice call travels through intermediate routers, and these routers should configure for local and remote dial peers if they have voice devices attached.

FXO INTERFACES CONFIRGURATION

The main function of FXO interfaces provide a gateway from the VOIP network to the analog PSTN, or to a PBX that does not support E&M signaling. So users can reach telephones and fax machines outside the VOIP network.
To create a local dial peer for an FXO interface, the destination pattern refers to outgoing calls, and including wild card in it, because the PSTN performs the switching.
The VOIP feature can also remove digits that we don't want to send to the PSTN, I supposed the user want to dial 7 to reach an outside line (analog PSTN), the configuration can be made as a following:
Philadelphia2 (config)# **dial-peer voice 201 pots**
Philadelphia2 (config-dial-peer)# **dest-pat +7..........**
Philadelphia2 (config-dial-peer)# **port 1/0/0**
To enable philadelphia1 router users to make calls over the philadelphia2 router's local PSTN, the following commands is used:
Philadelphia1 (config)# **dial-peer voice 701 voip**
Philadelphia1(config-dial-peer)# **dest-pat +8..........**
Philadelphia1(config-dial-peer)# **session-target ipv4:192.168.20.1**
Philadelphia2(config)# **dial-peer voice 601 pots**
Philadelphia2(config-dial-peer)# **dest-pat +8..........**
Philadelphia2(config-dial-peer)# **port 1/0/0**
The philadelphia1 router now sends all calls whose numbers begin with special prefix 8 over the ip network to the philadelphia2 router. the philadelphia2 router removes the 8 and passes the calls through its analog FXO gateway to the local PSTN.

E&M INTERFACES CONFIGURATION

If we have more than a few voice users at each location, then the cost of voice ports and routers, and the effort needed to configure dial peers for all the combinations of origins and destinations, increases rapidly. In this situation, it may be more efficient to use a PBX at each location to switch local traffic and to direct incoming calls, and to connect the PBXs over an IP network using E&M voice interface cards.
Consider the Philadelphia braches, now each branch has a PBX to operate its internal telephone network. While the IP network carries voice traffic between the branches. In the following figure, each BPX connects to the IP router over an E&M interface connection.
Figure (3) Linking PBXs over the IP Network (Local Dial Peers)



I supposed both BPX use E&M interface type 2, with four-wire operation and immediate-start signaling. The value used in coming configuration depends on PBX, and provided from the concern telecommunication department or the PBX manufacturer.

In the following configuration, philadelphia2 users can dial 8 and a four-digit extension to reach telephones in the philadelphia1 branch. Philadelphia1 users can dial 5 and a four-digit extension to reach telephones in the philadelphia2 branch

The philadelphia2 router is connected to the PBX over E&M port 3/0/0. That means that this port is associated with local (POTS) dial peers for incoming calls. But we no longer need to associate every telephone number with its own port. Instead, I can configure a local dial peer as if all the philadelphia2 telephones (represented by a wild-card destination pattern) are connected directly to this port, as shown in the following commands:
Philadelphia2 (config)# **dial-peer voice 111 pots**
Philadelphia2 (config-dial-peer)# **dest-pat +1408555....**
Philadelphia2 (config-dial-peer)# **port 3/0/0**

Remote (VoIP) dial peers for outgoing calls associate destination phone numbers on the philadelphia1 router with that router's IP address:
Philadelphia2 (config)# **dial-peer voice 121 voip**
Philadelphia2 (config-dial-peer)# **dest-pat +1919958....**
Philadelphia2 (config-dial-peer)# **session-target ipv4: 192.168.11.3**
Philadelphia2 (config-dial-peer)# **exit**
Philadelphia2 (config)#

Philadelphia2 (config)# **num-exp 8.... +1919958....**

To configure the E&M port similarly to any other network interface, using the follwing command:
Philadelphia2 (config)# **voice-port 3/0/0**
Philadelphia2 (config-voice-port)# **signal immediate**
Philadelphia2 (config-voice-port)# **operation 4-wire**
Philadelphia2 (config-voice-port)# **type 2**

To configure philadelphia1 router, the PBX for this router is connected to E&M port 2/0/1. the following commands configure a local dial peer for all philadelphia1 telephones:
Philadelphia1 (config)# **dial-peer voice 211 pots**
Philadelphia1 (config-dial-peer)# **dest-pat +1919958....**
Philadelphia1 (config-dial-peer)# **port 2/0/1**
The following commands configure dial peer for telephones on the philadelphia2 router:
Philadelphia1 (config)# **dial-peer voice 221 voip**
Philadelphia1 (config-dial-peer)# **dest-pat +1408555....**
Philadelphia1 (config-dial-peer)# **session-target ipv4: 192.168.16.1**
Philadelphia1 (config-dial-peer)# **exit**
Philadelphia1 (config)#
To configure number expansion, to make it easy for philadelphia1 users to dial numbers on the philadelphia2 router:
Philadelphia2 (config)# **num-exp 5.... +1408555....**
To configure the E&M port:
Philadelphia1 (config)# **voice-port 2/0/1**
Philadelphia1 (config-voice-port)# **signal immediate**
Philadelphia1 (config-voice-port)# **operation 4-wire**
Philadelphia1 (config-voice-port)# **type**
And to save all the configuration, to each router the following commands is used:
Press **Ctrl-Z**.
Router# **show running-config**
Router# **copy running-config startup-config**
Building configuration. . .
[OK]
Router#
now the router is configured to boot in the new configuration.

CONCLUSION

This study showed that call communications system providers would be able to easy install and deploy the VoIP technology where ever they need. As stated before, this technology will assist various types of organizations to provide more reliable, efficient, upgradeable, low-cost and secure way of communication.

Although few office environments and even fewer homes use a pure VoIP infrastructure, telecommunications providers routinely use IP telephony, often over a dedicated IP network, to connect between their switching stations, where they convert the dedicated voice signal to IP packets and back. The result is a data-abstracted digital network which the provider can easily upgrade and use for multiple purposes. Corporate customer support centers which provide support over telephone often use IP telephony exclusively to take advantage of the data abstraction that comes with it.

The benefit of using this technology is the need for only one class of circuit connection and better use of the available bandwidth. IP telephony is commonly used to route traffic that may be originated from and terminated at conventional PSTN telephones.

VoIP is now widely deployed by carriers, especially for international telephone calls. Most commonly, users are completely unaware that their telephone call is being routed over IP infrastructure for most of its distance, instead of entirely over the circuit switched PSTN.

This study will lead to the use of VoIP also by large companies to eliminate call charges between their offices, by using their data network to carry inter-office calls. They may also use VoIP to reduce the costs of calls outside the company, by carrying them to the nearest point on their network before handing them off to the PSTN by offering a gateway to the PSTN from any VoIP phone.

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APPENDIX

A. Rules and Regulations Regarding the Implementation of VoIP Technology

The statement is to be issued by the TRC in order to bring clarity and certainty to the regulatory treatment of Voice over Internet Protocol (VoIP) technologies. This issue has been complex for operators, service providers and users in Jordan, as VoIP technology tends to blur the lines between "voice" service and "data" service. This is problematic in part because the regulatory regime in Jordan, in particular the License of Jordan Telecommunications Company (Jordan Telecom) makes a clear distinction between "voice" and "data" services and treats them differently as a legal and regulatory matter. This statement does not attempt to define "voice" or "data" any more specifically than it already is defined, but rather attempts to clarify that the existing rules govern the activities of "service providers" and "users" differently.

The TRC has undertaken an extensive review of this issue, including consideration of the public consultation conducted by the Ministry of Information and Communications Technology (MoICT), as well as the treatment of this issue by telecommunications regulators in other jurisdictions.

The Statement's primary concern is the scope of the exclusivity currently enjoyed by Jordan Telecom for the provision of Public Switched Voice Service. This exclusivity is contained in its License and expires December 31, 2004.

This statement was released in draft form on August 14th and comments were invited from interested parties. Comments were received from BossIT, TEData, Jordan Telecom, Cyberia, Batelco Jordan and Khaled Hudhud. The TRC has taken all comments received into account in formulating its response [4].

B. FORMAL STATEMENT REGARDING VOIP

The TRC views the duty to foster the development of innovative telecommunications technologies in Jordan to be at the core of the strategic goals of the TRC and of Jordan. At the same time, the TRC is bound by the terms and conditions of the Telecommunications Law and the licenses and other regulatory instruments it has issued. It takes seriously its duty to ensure regulatory certainty by making its rulings as clear and consistent with one another and the Telecommunications Law as possible. Balancing these two duties is one of the TRC's most important challenges. The TRC therefore issues this formal statement to clarify the application of Jordan Telecommunications Company's license terms to the use of Voice over Internet Protocol technologies in Jordan.

1. The TRC views voice service provided using VoIP technology as the functional equivalent of voice service provided using circuit switched voice technology. Jordan Telecom therefore possesses the exclusive right to provide Public Switched Voice Service using VoIP technologies pursuant to Section 2.4 of its License.
2. Jordan Telecom's exclusive right to provide Public Switched Voice Service covers only the provision of a commercial service by one party to another separate party. Persons in Jordan are therefore free to utilize their legitimately subscribed data communication circuits to transmit and receive voice or other media format of their choosing without fear of infringing on Jordan Telecom's License, as long as they are transmitting and receiving such information for their own private use and not on behalf of a third party.
3. The use of VoIP technologies by commercial service providers other than Jordan Telecom to provide voice service to the public is prohibited prior to January 1, 2005. After that time, the provision of voice service using VoIP will be governed by future rules and/or guidelines to be issued by the TRC. The current prohibition includes services that utilize a normal telephone connected to the public switched telephone network, which then routes calls over the Internet via a gateway operated by a service provider, as well as the commercial offering of voice service to the public by the owner of a personal computer or other device connected to the Internet.
4. No entity other than Jordan Telecom may offer voice service to the public utilizing VoIP prior to January 1, 2005. The TRC views any activity by data licensees to market VoIP services to its users as equivalent to offering Public Switched Voice Services and therefore prohibited by Jordan Telecom's exclusive right to provide such services. The TRC therefore prohibits any activity by Data Communication Licensees to market voice service via VoIP to the public in Jordan using prepaid cards, hardware devices with included payment functionality or software.

5. The use of VoIP technologies prior to January 1, 2005 by commercial service providers other than Jordan Telecom to convey foreign-originated voice calls for termination on the public switched telephone network in Jordan, including on the network of mobile operators, constitutes an infringement of Jordan Telecom's License. The use of such technologies after December 31, 2004 will be subject to applicable TRC decisions to be adopted n the future.
6. Any infringement of Jordan Telecom's license by any entity providing commercial voice service using VoIP should be brought to the immediate attention of the TRC, which will take appropriate measures. The TRC will also take active measures to detect such activity and prosecute offending parties.
7. No Internet service provider, including Jordan Telecom, is authorized to block or otherwise interfere with the activities of any Internet user in Jordan in accessing or communicating with users or other entities outside of Jordan using VoIP without express prior authorization from the TRC.
8. Any person who provides voice service to the public utilizing VoIP in contravention of this decision or of the License of Jordan Telecom will be subject to the sanctions of the law including Articles 62-64 and 78-79 of the Telecommunications Law. Such enforcement action may result in seizure of equipment, financial penalties, prison terms or a combination of the above. The TRC will endeavor to apply the strictest penalties available in the event of a violation of these provisions.